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Duration modeling with Hidden Markov models

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PROGRAM OF

The 126th Meeting of the Acoustical Society of America

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NOTE: All Journal articles and Letters to the Editor are peer reviewed before publication. Program abstracts, however, are not reviewed before publication, since we are prohibited by time and schedule.

MONDAY AFTERNOON, 4 OCTOBER 1993

TERRACE ROOM, 12:55 TO 4:45 P.M.

Session 1pAO

Acoustical Oceanography: Acoustical Determination of Polar Ocean Processes I

Subramaniam D. Rajan, Cochair

*Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution,
Woods Hole, Massachusetts 02543*

James H. Miller, Cochair

Department of Electrical and Computer Engineering, Naval Postgraduate School, Monterey, California 93943

Chair's Introduction—12:55

Invited Papers

1:00

1pAO1. Fracture of sea ice. Ira Dyer (Dept. of Ocean Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

The major cause of ambient noise in the Arctic Ocean is radiation from the fracture of sea ice in response to environmental forcing. Extant noise data on fracture in pack ice in the central Arctic and in free-drifting ice at its margins, including measures such as spectral density, temporal evolution, and individual ice fracture transients and their statistics are reviewed. Physical mechanisms responsible for ice fracture are of fundamental interest, and therefore the fracture mechanics of sea ice, which leads to a picture of ice failure not unlike that of failure in the Earth's crust via earthquakes, is also discussed. As in earthquakes, ice fracturing is a process of many small fault plane motions which can result in a major ice crack. Prospects for use of noise measurements to understand ice processes immediately come to mind, as well as their use intrinsically to describe acoustic properties of the Arctic Ocean.

1:25

1pAO2. Recent approaches to the acoustic-seismic study of ice mechanics. David M. Farmer and Yunbo Xie (Inst. of Ocean Sci., P. O. Box 6000, Sidney, BC V8L 4B2, Canada)

Acoustic and seismic emissions provide useful information on the mechanical properties of sea ice and its response to stress. Two recent field studies carried out under the ONR Ice Mechanics initiative have provided opportunities for exploring this approach to remote sensing using both passive and active measurements. Passive sensing revealed significant spatial variability in the response of the ice to thermal stress. Detailed analysis of the acoustic signal in the water yields insight on the source mechanism, which included local uplift, subsequently verified by inspection of the fracture site. Flexural waves in the ice were anisotropically distributed with blocking by a nearby pressure ridge dominating the spatial variability. Variations in ice properties were also probed with artificial impacts on the ice surface, revealing significant variability in both shear and p -wave propagation. The influence of the pressure ridge was very marked in its attenuation of flexural waves, but significant modifications of shear and p -wave velocities were also measured. Finally, advantage was taken of an artificially induced fracture experiment conducted by Dempsey and Spencer in April 1993. Controlled fracturing was induced with a flat-jack leading to multiple acoustical and seismic

emissions that were detected with a hydrophone array. Such measurements should provide a basis for comparison with models for the fracture of heterogeneous materials.

1:50

1pAO3. Oceanographic inferences from ambient noise in the marginal ice zone. Michael J. Buckingham (Marine Phys. Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0213)

In the polar seas the spatial structure of the ambient noise is often determined by the upward refracting sound-speed profile. Such a profile supports normal modes, even in the absence of bottom interactions. A full theoretical (analytical) model of ambient noise in an inverse-square profile—representative of polar waters—has been developed [Buckingham, *Philos. Trans. R. Soc.* (in press)], which yields expressions for the auto and cross spectral densities of the noise that are valid at infrasonic frequencies (VLF) and low frequencies (LF) up to several kilohertz. Three terms appear in these expressions, representing the modal and continuous components of the noise. When the theoretical spectra are compared with ambient noise data taken in the marginal ice zone off the east coast of Greenland, a crude distribution of noise sources at the ice edge can be inferred and bottom effects in the observed spectra become apparent. The spatial inhomogeneity of the noise is also very evident from the theory, which has implications concerning beamforming for obtaining noise directionality information. [Research supported by ONR.]

2:15

1pAO4. Ultrasonic modeling of sea ice seismoacoustic wave phenomena. Jacques R. Chamuel (Sonoquest Advanced Ultrason. Res., P. O. Box 153, Wellesley Hills, MA 02181-5339)

A variety of broadband Arctic seismoacoustic wave phenomena are demonstrated using laboratory ultrasonic modeling techniques introduced by the author in the mid 1970s to study the generation, propagation, detection, and scattering of low-frequency underwater acoustic waves coupled to the heterogeneous ice cover. Data on frequency-dependent under-ice reflectivity, effective elastic properties, low-frequency transmission loss, horizontal refraction, wave conversion, and low grazing angle backscattering are discussed. Different guided leaky and trapped waves from range-dependent models are isolated and examined including flexural waves, shear horizontal waves, Rayleigh waves, Scholte waves, edge plate waves, and wedge waves. The laboratory models consisted of finite floating plates with cracks, ridges, low-velocity layers, boreholes, trapped air pockets, solid wedges, and suspensions. The results provide physical insight into realistic 3-D problems and lead to the development of comprehensive understanding of near-field and long-range seismoacoustic phenomena essential for interpreting and inverting Arctic acoustic data, planning field experiments, and predicting ice failure modes from acoustic emission. The generic findings are applicable to ocean bottom interacting acoustic waves, nondestructive testing, and ultrasonic imaging. [Work supported by DREP and ONR.]

2:40

1pAO5. Feasibility of acoustic thermometry of Arctic Ocean climate change. Peter N. Mikhalevsky (Sci. Appl. Int. Corp., Ocean Sci. Group, McLean, VA)

Inverting observations of changes in long-range acoustic travel time, phase, amplitude, and modal dispersion has been proposed for the Arctic Ocean. Initial calculations [Mikhalevsky *et al.*, "Arctic Ocean Warming: Can We Measure it Acoustically?," *The Oceanographic Society, Second Scientific Meeting, St. Petersburg, FL (March 1991)*] showed that it could be possible to monitor ocean volume warming, and changes in sea ice extent and thickness. New results will be presented using higher fidelity acoustic modeling and the results of other international collaborators, including the Russians, will be summarized. Plans for a feasibility test in the Arctic are underway for a spring 1994 experiment.

3:05–3:15 Break

Contributed Papers

3:15

1pAO6. Monitoring thermal fracturing in ice: The relation to ice stress and strength. P. J. Stein (Scientific Solutions, Inc., 47 Aster Ct., Nashua, NH 03062), J. K. Lewis (Atlantic Appl. Res. Corp., Long Beach, MS 39560), and W. B. Tucker (U. S. Army Cold Regions Res. and Eng. Lab., Hannover, NH 03755)

Here observations of thermally induced ice fracturing in the Arctic are summarized. Thermally induced fractures may be an important phenomena in understanding ice strength. Sets of observations of event locations using hydrophone and geophone arrays along with meteorological and ice stress measurements are discussed. Data are reviewed for both first and multiyear ice floes. Stress observations show that smooth first year ice can support much higher tensile forces. The event location data indicate that indeed first year ice is much less susceptible to thermal fracturing. This indicates that smooth first year ice is in some manner stronger than multiyear ice.

3:30

1pAO7. Acoustical determination of the ice growth in the Arctic. Henry A. Laible (Dept. of Ocean Eng., MIT, Cambridge, MA 02139 and Dept. of Appl. Ocean Phys. Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543) and S. Rajan (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

A crosshole tomography experiment was conducted in the Arctic between April 1992 and March 1993 to determine the time evolution of the acoustic/elastic properties of sea ice. Data were taken every third day for 9 months using an array of vertical and horizontal transducers located in the ice and water column below the ice. A method to monitor the growth of the underside of the ice consisted of comparing the arrival times of both the direct path and reflected path from transmitters in the water column to the receivers in the water column. The differences in arrival times over the 9 months indicates as much as 1-m growth of ice. In addition to the ice growth estimates, the reflection coefficient was determined as a function of time and angle. Acoustic characteristics of

3:45

1pAO8. Measurement of large scale deformation of the Arctic ice pack using acoustic backscatter. Greg Duckworth, Kevin LePage, and Ted Farrell (Bolt Beranek and Newman, Inc., 70 Fawcett St., Cambridge, MA 02138)

The ONR AEAS Program sponsored a low-frequency reverberation study program in the Central Arctic as part of the AREA-92 Exercise in March–April of 1992. The ZIRCON ice camp was the site of the monostatic and bistatic active sonar experiment with acoustic sources, a 256-channel billboard receive array, and real-time processing and display system. The reverberation data collected have been coherently processed between sequential pings to estimate the bearing, range, and radial velocity of ice scattering footprints of various range and bearing extents. Due to the highly repeatable nature of reverberation from the ice canopy, high source strength, and the gain of the receiver array, this processing is shown to be capable of unambiguously estimating the movement of these footprints over a 250-km-radius area to better than 10-m accuracy for transmissions as far apart as 4 h. Strain rate estimates using the technique agree with sparse GPS data in the area covered, and are shown to be well modeled by a simple three-parameter strain rate field. The details of the output from the processing also shed light on the fundamental acoustic propagation and backscatter processes themselves. [Work supported by ONR.]

4:00

1pAO9. Arctic ambient noise and climate change. Warren W. Denner (EOS Res. Associates, P. O. Box 1378, Carmel Valley, CA 93924)

In global warming, sea ice plays a significant role, and sea ice conditions may provide one of the most sensitive parameters to monitor. Modeling predicts that warming will be greatest at high latitude, and sea ice properties are very sensitive to warming and changes in atmospheric forcing. Therefore, sea ice might provide a sensitive marker for climate change. The mechanical properties of sea ice are related to the ice temperature, salinity, thickness, concentration, and flow size distribution. It has been established that the ambient noise field under pack ice is correlated to the mechanical properties, ice conditions, and forcing. Many studies over the last two decades have delineated some of these relationships. The hypothesis of this paper is that monitoring the ambient noise under the ice, for which there is well-developed and reliable technology, may be an excellent way to detect a climate change signal. This paper will review the relationships between under ice ambient noise and the properties of the ice pack. Based on what is known, a monitoring system consisting of remotely operated buoys is proposed. A proposed buoy design and sampling array for the Arctic is presented. Used in conjunction with satellite remote sensing, the system could provide for continuous monitoring of the ice pack for other purposes and would be cost effective for acoustic monitoring of the climate change signal in the Arctic.

1pAO10. Relating Arctic ambient noise to thermally induced fracturing of the pack ice—An ability to monitor climate change? James K. Lewis (Atlantic Appl. Res. Corp., 207 S. Seashore Ave., Long Beach, MS 39560)

Thermally induced fracturing of pack ice can be simulated using a viscoelastic model and a fracturing paradigm for predicting the stress state of the ice. Under a given set of circumstances, the fracturing paradigm simulates conditions that exceed the tensile strength of the ice as well as the associated stress relief as a result of fracturing. The thermal stress–ice fracturing model was used to produce a fracture count time history for the ice pack during the fall of 1988 in the eastern Arctic Ocean. The count was compared to the observed 500-Hz under-ice noise in which 13 distinct noise episodes were identified. The simulation showed increases in predicted fracturing levels for 11 of the 13 noise episodes. Some of the fracturing was associated with nightly cooling. Others occurred as a result of sustained cooling over several days following the passage of cold fronts. The stress–fracturing model can be used to study changes in under-ice noise as a result of global warming. Examples are presented based on warming as predicted by various climate models. The results indicate that climate changes could be monitored using winter-time under-ice ambient noise observations from the Arctic.

4:30

1pAO11. Oceanological aspects of acoustic monitoring of Arctic Ocean climate. K. D. Sabinin (N. N. Andreyev Acoust. Inst. and Ocean Acoust. & Inform. Ltd., 38 Vavilov St., Moscow, 117942, Russia) and S. V. Pisarev (State Oceanogr. Inst. and Ocean Acoust. & Inform. Ltd., Moscow, 117942, Russia)

The Arctic Ocean is an important object of investigation in the global ATOC program because of its crucial role in the heat balance of the Northern Hemisphere. Although the temporal and spatial variability of the sound speed in the Arctic Ocean is relatively weak, the Arctic ice cover essentially complicates modeling of the acoustic response to climate change. Moreover, the influence of the strongly varying ice cover causes additional difficulties in interpretation of the results of acoustic ocean thermometry for several reasons: (1) the extremely strong influence of ice distribution on the ocean–atmosphere heat exchange, (2) the long-term variation of the ice cover, and (3) the specific response of the surface water sound speed to heating of an ice-covered sea (a heat input should decrease the sound speed under melting ice). Under-ice winter convection can also create specific kinds of inhomogeneities of the Arctic Ocean (chimneys, plumes of dense water, etc.). Additional serious difficulties arise because of strong dependency of Arctic climate on the heat inflow of both the Atlantic and Pacific Oceans. This changeable inflow can cause long-term variations of the Arctic Ocean. It seems worthwhile to begin the Arctic ATOC with the installation of acoustic tracks across the Fram Strait.

Session 1pSP

Speech Communication: Models and Mechanisms in Speech Production

Ronald C. Scherer, Chair

The Denver Center for the Performing Arts, 1245 Champa Street, Denver, Colorado 80204

Contributed Papers

12:30

1pSP1. Voice simulation with a three-mass model of the vocal folds. Brad H. Story and Ingo R. Titze (Natl. Ctr. for Voice and Speech, Dept. of Speech Pathol. and Audiol., Univ. of Iowa, Iowa City, IA 52242)

The classic two-mass model of the vocal folds [K. Ishizaka and J. L. Flanagan, *Bell Syst. Tech. J.* **51**, 1233–1268 (1972)] has been extended to a three-mass model in order to represent more accurately the body-cover vocal fold structure [M. Hirano, *Folia Phoniat.* **26**, 89–94 (1974)]. The model consists of two "cover" masses that are coupled to a "body" mass by nonlinear springs and viscous damping elements. The body mass, which represents muscle tissue, is also coupled to a rigid wall (assumed to represent the rigid laryngeal framework) by a nonlinear spring and damping element. The two cover springs are intended to represent the vocal ligament tension while the body spring simulates the tension produced by contraction of the thyroarytenoid muscle (TA). Additionally the two cover masses are coupled to each other through a linear spring that can be regarded as representing the surface tension of the cover and hence largely contributes to the mucosal wave velocity. Effects on the vocal fold movement due to changes in subglottal pressure, spring tensions, element masses, and vocal tract loading will be shown.

12:45

1pSP2. Analysis of rough voice from a nonlinear dynamics point of view. David A. Berry, Ingo R. Titze (Natl. Ctr. for Voice and Speech, Dept. of Speech Pathol. and Audiol., Univ. of Iowa, Iowa City, IA 52242), and Hanspeter Herzel (Humboldt University, O-1040 Berlin, Germany)

Because the vocal folds are nonlinear systems with many degrees of freedom, bifurcations and chaos should be expected for certain parameter configurations. Indeed, bifurcations and chaos appear in some of the simplest models of vocal fold vibration. They are also seen in more complex models, in models that incorporate left–right asymmetries, and in biomechanical simulations of vocal fold oscillations. Furthermore, an acoustical analysis of many types of rough voice (e.g., creaky voice, vocal fry, and newborn infant cries) reveals an intimate relationship between voice mechanics and bifurcations and chaos. Likely physiological mechanisms of the observed nonlinear phenomena are presented. Basic concepts and analysis techniques from nonlinear dynamics are reviewed and compared with traditional voice analysis schemes. Finally, it is shown how much of the terminology in the literature describing rough voice can be unified within the framework of nonlinear dynamics. [This research was supported by Grant No. P60 DC00976 from the National Institute on Deafness and Other Communication Disorders.]

1:00

1pSP3. Phonation threshold pressure in a physical model of the vocal folds. Ingo R. Titze and Michael R. Titze (Natl. Ctr. for Voice and Speech, Dept. of Speech Pathol. and Audiol., Univ. of Iowa, Iowa City, IA 52242)

A physical model of a vocal fold was constructed with three layers: an artificial epithelium (latex), a superficial Reinke's space (water), and an immobile vocal fold body (aluminum). This vocal fold was positioned in a plexiglas airway such that the glottal aperture and the convergence angle could be varied systematically with respect to a solid boundary, which represented the opposite vocal fold in a hemilarynx configuration. Subglottal pressure was controlled with a constant pressure valving system. Phonation threshold pressure (the Hopf bifurcation in nonlinear dynamics) was measured as a function of glottal aperture and divergence angle. This pressure increases with increased aperture and convergence, as predicted by theory, but results for a divergent glottis are not as easy to interpret. [This research was supported by Grant No. P60 DC00976 from the National Institute on Deafness and Other Communication Disorders.]

1:15

1pSP4. Noninvasive phonation threshold pressure estimates: Some methodological considerations. Kimberly Fisher (Univ. of Oklahoma Health Sci. Ctr., 825 N.E. 14th St., Oklahoma City, OK 73190)

Phonation threshold pressure (PTP) is the minimum subglottal pressure required to initiate vocal fold oscillation. This pressure is important both theoretically and clinically because of its relationship to tissue viscosity, time varying glottal flow, and vocal tract acoustics. The effects of several different measurement methods on estimated PTP were studied in normal subjects. Oral pressures during quietly voiced consonant–vowel–consonant syllable strings were used to estimate PTP. Factors expected to impact PTP estimates included: (1) low nasal flows associated with soft, relaxed phonation, and (2) vowel context, due to possible reactive loading. A significant percentage of normal subjects evidenced low, inconsistent nasal airflow during /p/ occlusion near threshold (as indicated by nasal pneumotachography). Nasal flow was not associated with supra-threshold production for these same subjects. In spite of similar instruction and pre-recording practice, subjects varied in the strategies they used to accomplish phonation at soft levels near their threshold pressure. Strategies potentially involving slight adjustment in glottal adduction were observed and would be predicted to alter PTP. These methodological challenges will be discussed relative to the variability observed in PTP estimates, and the potential impact on validity and reliability of noninvasive oral pressure measurement.

1:30

1pSP5. Laryngeal resistance of an *in vivo* canine model of phonation with a constant air pressure source. Sina Nasri, Ali Namazie, Ming Ye, Joel A. Sercarz, Jody Kreiman, Bruce R. Gerratt, and Gerald S. Berke (Div. of Head and Neck Surgery, UCLA School of Medicine, 10833 Le Conte Ave., Los Angeles, CA 90024-1624)

Previous studies of laryngeal biomechanics using *in vivo* models have generally used a constant air flow source. Several authors have recently suggested that during phonation, the lung–thorax system functions as a constant pressure source. The present paper describes an *in vivo* canine system designed to maintain a constant peak subglottic pressure (P_{sub}) using a pressure-controlling mechanism. It was found that with a de-

crease in P_{sub} , the range of recurrent laryngeal nerve stimulation (RLNS) voltage needed to induce phonation was reduced. At a given superior laryngeal nerve stimulation (SLNS) level and P_{sub} , increasing levels of RLNS resulted in a normal distribution of vocal efficiencies. For each SLNS and P_{sub} , minimum and maximum levels of RLNS were determined outside of which no phonation was possible. Levels of RLNS that produced an optimal stable phonation were also identified. Increasing levels of RLNS resulted in significant decreases in glottal air flow. Contrary to a previous report using a constant flow source, increasing levels of SLNS produced a significant decrease in glottal resistance. This is consistent with another previous study [D. M. Moore and G. S. Berke, *J. Acoust. Soc. Am.* **83**, 705-714 (1988)] demonstrating that the open quotient increased with increasing SLNS.

1:45

1pSP6. A study of pulsatile airflow in the excised larynx model. Fariborz Alipour (Dept. of Speech Pathol. and Audiol., Univ. of Iowa, Iowa City, IA 52242-1012) and Ronald C. Scherer (Denver Ctr. for the Performing Arts, Denver, CO)

The pulsatile airflow in the excised larynx was investigated with the simultaneous measurements of glottal pressure, volume flow, EGG, and particle velocity at various dynamic conditions of the larynx. Canine larynges were mounted in a bench with some sutures attached to cartilages to mimic the function of laryngeal muscles. The sustained oscillations were obtained and maintained with the flow of heated and humidified air through the trachea. The pitch and amplitude of oscillations were controlled with amount of airflow and the mechanical adjustment of glottal adduction and vocal-fold elongation. The particle velocity was measured above the glottis with a constant temperature hot-wire probe at various locations and oscillating conditions. The measured subglottal pressure with a pressure transducer had a periodic variation and its mean value was higher than the manometer's reading of the same location. The measured particle velocity showed some turbulence superimposed on a periodic flow. The mean particle velocity showed some variations in the glottal exit area both along the vocal folds and across the glottis. [Work supported by NIDCD Grant No. DC00831-02.]

2:00

1pSP7. The effect of gas density on glottal vibration and exit jet particle velocity. Steven Bielamowicz, Gerald S. Berke, Jody Kreiman, Bruce R. Gerratt, David C. Green (UCLA Div. of Head and Neck Surgery, UCLA School of Medicine, CHS 62-132, Los Angeles, CA 90024 and VA Med. Ctr., West Los Angeles, CA 90073), and Richard S. McGowan (Haskins Labs., New Haven, CT 06511)

Although theoretical studies include a term for gas density in their mathematical descriptions of glottal aerodynamics, no studies have measured the effect of gas density on glottal vibration and particle velocity. This study used a constant temperature anemometer in the *in vivo* canine model of phonation to evaluate the effect of gas density on subglottic pressure, particle velocity, and glottal vibration by comparing phonation with air and helium. With gas flow and laryngeal nerve stimulation held constant, peak subglottic pressure was significantly greater during helium phonation (70 cm Hg) than during air phonation (62 cm Hg). In addition, peak particle velocity during helium phonation (45 m/s) was significantly greater than phonation with air (34 m/s). However, the increase in particle velocity with helium compared to air was less than predicted by the Bernoulli relationship. This loss in velocity likely represents turbulent and frictional forces at the glottal outlet.

2:15

1pSP8. The role of importance sampling in the establishment of normal ranges for speech characteristics. Jon H. Lemke and Hani Samawi (Dept. of Preventive Med. and Environ. Health, 2800 Steindler Bldg., Univ. of Iowa, Iowa City, IA 52245)

Most measurements of voice characteristics have inherently skewed distributions that have never been characterized for individuals without

speech disorders, that is, individuals with "normal" voices. Any diagnostic value these measures might have requires that one studies the characteristics of these "normal" voice distributions, in particular the quantiles necessary to establish the normal range. The focus will be on perturbations measures, such as jitter and shimmer. To solve this problem a method of importance sampling has been developed for estimating quantiles of skewed distributions. Importance sampling is a modified bootstrap procedure with exponential tilting, that is, a resampling method where the probabilities of the original observations appearing in the new samples are no longer equally weighted but are weighted by their closeness as an order statistic to the quantile being estimated. To estimate the normal limits of jitter and shimmer without resampling procedures will require at least 800 subjects. However, to estimate the normal limits with the same efficiency using this modified importance sampling technique will require approximately 80 subjects. The assessment of voice is very expensive and time consuming. There will never be 800 normal voice assessments conducted under identical conditions by anyone. How modified importance sampling can be used to establish normal limits for jitter and shimmer will be demonstrated with data collected at the Denver Center for the Performing Arts, Denver, Colorado, and the Wendell Johnson Speech and Hearing Center, University of Iowa, Iowa City, Iowa.

2:30-3:00 Break

3:00

1pSP9. Speech production: Thirty years after. Arthur S. House (IDA Ctr. for Commun. Res., Princeton, NJ 08540) and Kenneth N. Stevens (MIT, Cambridge, MA 02139)

Measurements have been made on a corpus of bisyllabic nonsense utterances spoken by three adult male talkers on two occasions separated by 30 years. The 1960 and 1990 recordings were processed similarly so that comparisons could be made directly. Some primary and secondary characteristics of vowels were measured: duration, fundamental frequency and the frequencies of $F1$ and $F2$. The durations of prestressed consonants were measured, and vowel-consonant amplitude ratios computed. Long-term spectra of the talkers were examined and some informal listening comparisons were made. Spectrograms of some samples were studied, and statistical analyses, when appropriate, were made. Results indicate that the talkers' productions were remarkably consistent over 30 years. Over time the talkers' mean vocalic duration (10 vowels) shortened (6 ms); their mean vocalic fundamental frequency rose (7 Hz); their mean frequencies of $F1$ and $F2$ (10 vowels) descended (15 and 26 Hz, respectively); their mean (initial) fricative duration lengthened (12 ms); and, their mean (initial) stop duration lengthened (hold portion, 15 ms; release portion, 5 ms). Interactions were usually significant: individual talkers' changes in duration, formant frequencies; and fundamental frequency were not identical. Patterns of duration and fundamental frequency in the set of vowels, however, were stable over time.

3:15

1pSP10. Motor equivalence in the transformation from vocal-tract configurations to the acoustic transfer function: Adaptation to a bite block. J. S. Perkell, M. L. Matthies, M. A. Svirsky, and M. I. Jordan (Speech Commun. Group, Res. Lab. of Electron., MIT, Cambridge, MA 02139)

In a preliminary study [Perkell *et al.*, *J. Acoust. Soc. Am.* **93**, 2948 (1993)] subjects demonstrated a trading relation (negative correlations) between tongue-body raising and lip protrusion in production of the vowel /u/. Data analyses indicated an increased tendency for the trading relation to occur at high values of $F2$. The tentative hypothesis is that when more than one articulator contributes to producing a particular acoustic cue and a source of variation displaces one of the articulators away from its target, compensating (complementary) adjustments are made in the programmed displacements of the other contributing articulator(s) to help constrain the acoustic variation.

These adjustments should be more strongly expressed in tokens that are close to phonemic boundaries, where variation in the displacement of one of the contributing articulators could result in production of the wrong sound. This work has been extended with two new subjects. halfway through the experiment a bite block was placed in the subject's mouth to introduce an additional source of variation. One subject strictly used coordination (i.e., positive correlation) to produce /u/ and did not alter this strategy with the bite block. Negative correlations were found for the other subject in subsets with relatively high values of $F2$. [Work supported by NIDCD.]

3:30

1pSP11. Formant sensitivity in the vocal tract. Sungbok Lee (Central Inst. for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Sensitivity of formants to perturbations of the location and degree of tongue constriction was studied. These two tongue parameters were extracted from vowel-like midsagittal tongue shapes ($N=321$) generated using a tongue model [Harshman *et al.*, *J. Acoust. Soc. Am.* **62**, 693–707 (1977)]. The tongue shape data were arranged such that the tongue parameters vary monotonically. Lip opening was varied from 0.3 to 6.0 cm² in 0.3-cm² steps and the first three formants of each vocal tract were computed. Formant sensitivity was evaluated as dF/dA (Hz/mm), where dA is perturbation in a tongue parameter and dF is the corresponding formant change, and was plotted as a function of the location of tongue constriction. The results can be interpreted in terms of the formant variability, or stability, of a given vocal tract configuration. One finding is that at the tongue position associated with the vowel /u/ the formants are very stable under perturbation of the location of constriction. However, the same position is found to be the most unstable position for both $F2$ and $F3$ under perturbation of the degree of constriction. This may explain the largest $F2$ and $F3$ variations of /u/ in a set of normalized production data of vowels [Syrdal and Gopal, *J. Acoust. Soc. Am.* **79**, 1086–1100 (1986)]. [Work supported by NIDCD, DC 00296.]

3:45

1pSP12. Gradation of jaw perturbation effects on mandibular, labial, and velar kinematics. H. Betty Kollia (City Univ. of New York, New York, NY 10036 and Haskins Labs., 270 Crown St., New Haven, CT 06510)

It was recently shown that anatomically linked and remote articulators such as the jaw, the lips, and the velum are functionally constrained during bilabial closing, exhibiting temporal stability in the patterns of interarticulatory cohesion [Kollia *et al.*, *J. Acoust. Soc. Am.* **92**, 2390 (A) (1992)]. Further, it was found that, when functionally related for speech, jaw, lips, and velum show compensatory kinematic adjustments to a mechanical jaw perturbation, such as increased movement displacement and oral closing velocity, thereby maintaining interarticulator cohesion [Kollia *et al.*, *J. Acoust. Soc. Am.* **91**, 2474 (A) (1992)]. The interpretation is that speech movements are controlled in a global manner, rather than independently, reflecting large scale vocal tract actions. The present study aims at untangling the gradient effects of the perturbation on the articulators and their kinematic parameters. Overall load effects depended on the distance of the articulator from the site of the perturbation. Load effects were heightened in structures that were more proximal to the perturbation such as the jaw and the lower lip, and diminished in more remote structures, such as the upper lip and the velum. Moreover, load effects depended on the load onset time and had a variable effect on the different kinematic parameters. [Work supported by NIH Grant Nos. DC-00121 and DC-00594 to Haskins Laboratories.]

4:00

1pSP13. Inferring articulator positions from acoustics: An electromagnetic midsagittal articulometer experiment. John Hogden, Anders Lofquist, Vincent Gracco, Kiyoshi Oshima, Philip Rubin, and

Elliot Saltzman (Haskins Labs., 270 Crown St., New Haven, CT 06511)

To examine the mapping from acoustics to articulation, simultaneous articulatory and acoustic measurements were made of 90 vowel-to-vowel transitions produced by a Swedish speaker. Articulator positions were measured using an electromagnetic midsagittal articulometer that tracked seven receiver coils placed on the lips, jaw, and tongue. The vocal tract transfer function associated with each articulator position was estimated using cepstral analysis on short windows (25.6 ms) of the acoustic signal. The transfer functions were then vector quantized, giving each articulator configuration a corresponding vector quantization code. Given an acoustic signal and its corresponding vector quantization code (code 1 for example), an estimate of the position of any coil during the production of a speech signal can be made by averaging all the positions the coil assumed during the production of sounds labeled with vector quantization code 1. This produces a codebook that allows articulator positions to be estimated from acoustic signals. On a data set not used for training, correlations between estimated and actual receiver coil positions vary strongly with the place of attachment, but correlations above 0.9 are common for coils on the tongue. [Work supported by NIH.]

4:15

1pSP14. Duration aspects in manner of articulation distinctions. Joaquín Romero (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Differences in manner of articulation—between stops, fricatives, and approximants—have been described articulatorily in terms of degree and area of the constriction. It is suggested here that such differences are generally accompanied by a distinction in the duration of the constriction gestures. An experiment was carried out in which an optical tracking device was used to record the movement of the lips of a speaker of Andalusian Spanish, a dialect spoken in Southern Spain that has a complete series of homorganic stops, fricatives, and approximants. Results of both acoustic and kinematic analyses indicated that, while certain variation in constriction degree was indeed observed, the durational differences appeared to be much more stable and robust. A bilabial fricative [β] not only has a narrower constriction than an approximant β but is also longer. Fricatives, because they need to generate turbulent airflow, have a longer, more precise articulatory configuration than approximants, which create no turbulence and, consequently, need not remain in a particular articulatory configuration for long. Because of their very short duration, approximants seem to lack a precise articulatory target: they show a great deal of variation in their articulatory trajectories and are very sensitive to coarticulation from neighboring segments. [Work supported by NIH Grant Nos. DC-00121 and HD-01994 to Haskins Laboratories.]

4:30

1pSP15. The 3-D-tongue FEM model revised. Reiner Wilhelms-Tricarico (ATR International, Kyoto, Japan) and Chao-Min Wu (Biomed. Eng. Ctr., Ohio State Univ., Columbus, OH 43210-1002)

As a first application of a newly developed object-oriented finite element code that is specialized for the modeling of biological tissues we have been reimplementing the geometry of the tongue model by S. Kirtani *et al.* ("A computational model of the tongue," *Ann. Bul. Res. Inst. of Logopedics and Phoniatrics, Univ. Tokyo*, Vol. 10, pp. 243–251). The new implementation uses large strain and dynamic modeling of the tissue, includes the effects of inertia and observes precisely a weak formulation of the incompressibility constraint of the tissue. The muscle activation input for the current model is made similar to the original computation. However, a different, more realistic nonlinear muscle model is used, in which the active stress depends on the instantaneous strain and strain rate in the direction of the muscle fibers. The implementation methods and computational results will be presented. [Work supported by the Whitaker Foundation and by ATR-International.]

1pSP16. MRI measurements and acoustic investigation of the nasal and paranasal cavities. Jianwu Dang, Kiyoshi Honda (ATR Human Information Processing Res. Labs., 2-2 Hikaridai Seikacho Soraku-gun, Kyoto, 619-02 Japan), and Hisayoshi Suzuki (Shizuoka Univ., 3-5-1 Jouhoku, Hamamatsu, 432 Japan)

Morphological measurements of the nasal and paranasal cavities were performed to speculate on the acoustic properties of the human nasal tract. The magnetic resonance imaging (MRI) technique was used to measure the three-dimensional structure of the vocal tract. The area function of the nasal tract was calculated from seven subjects at rest. The whole vocal tract was measured from five subjects during

sustained pronunciation of nasal consonants. A marked morphological difference was observed between these data and the previous data particularly in the middle portion of the nasal tract. The previous data measured from cadaver specimens have a wide cavity in the middle portion possibly due to absent or dehydrated mucous membrane, while these data show narrow passages caused by thick mucosa. It was confirmed by an experiment that the above difference was reproducible by applying an adrenaline-like agent onto the nose. The acoustic transfer function was calculated from these data, and the speech sound, recorded soon after MRI scan, was used to evaluate the transfer function. The results indicate that the asymmetry of two nostrils can cause an extra pole-zero pair, and suggest that the paranasal cavities can play an important role in shaping spectral characteristics of human nasal cavities.

MONDAY AFTERNOON, 4 OCTOBER 1993

COLUMBINE ROOM, 12:55 TO 4:15 P.M.

Session 1pUW

Underwater Acoustics: Scattering and Reverberation I

Dale D. Ellis, Chair

SACLANT Undersea Research Center, Viale San Bartolomeo 400, 19138 La Spezia, Italy

Chair's Introduction—12:55

Contributed Papers

1:00

1pUW1. Scattering from under-ice features and estimation of ice roughness parameters. Tarun K. Kapoor and Henrik Schmidt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Previous results by Kapoor and Schmidt [J. Acoust. Soc. Am. **93**, 2419 (A) (1993)] have shown the possibility of isolating scattering hot spots under the ice sheet by matched-field processing of short-range returns assuming point scatterers. A natural extension of this approach is to model the scattered field, or replica field, assuming a more realistic model of the scatterers. Here, examination of the low-frequency scattering data from the CEAREX 89 field experiments conducted in the Central Arctic region with 1.8-lb SUS charges detonated at nominal depths of 800 ft is continued. The beamformer outputs from the crossed horizontal array and vertical line arrays are combined simulating a volumetric array, which provides much better resolution of the under-ice features. The scatterers are modeled as hemispherical and half-cylindrical protuberances. Importance of elastic properties of the ice and the protuberances will be addressed. Ice roughness parameters are estimated using the method of small perturbations. [Work supported by ONR.]

1:15

1pUW2. Retroreflective backscattering of sound in water due to Lamb waves on plates with corners: Observations. S. S. Dodd, C. M. Loeffler, and P. L. Marston (Appl. Res. Lab., Univ. of Texas, Austin, TX 78713-8029)

A leaky Lamb wave is known to be launched on a flat elastic surface plate in water when the surface normal lies near a cone whose symmetry axis gives the k vector of the incident sound. These water tank experiments with a 600-kHz sonar demonstrate that when the plate has a corner with edges meeting at angles of 90° and 45°, the wave vector of the Lamb wave is reversed due to repeated reflections at the edges that form the corner. The resulting leaky radiation gives a pronounced enhancement of the backscattering that depends only weakly on the orientation of the corner. Reductions in amplitude with deviations of the corner angle from 90° and with tilts of the surface normal were observed

and they generally support the approximate analysis of P. L. Marston *et al.* (Abstract 4pSA2 at this meeting). For a suitably cut randomly oriented plate, the retroreflective enhancement is more likely to be observed than the specular reflection since then the normal must lie on a narrow range of angles.

1:30

1pUW3. Long-range measurements of the strength and spectral character of low-frequency surface reverberation. Roger C. Gauss, Raymond J. Soukup, David M. Fromm (Naval Res. Lab., Washington, DC 20375-5350), and Joseph M. Fialkowski (Planning Systems, Inc., McLean, VA 22102)

Long-range monostatic measurements of low-frequency (185–985 Hz) and low-grazing-angle (nominally 5–15 deg) acoustic surface scattering were made in the Gulf of Alaska in February of 1992. The upward-refracting sound-speed profiles permitted ensouffication of the sea surface at ranges of several tens of kilometers without interacting with the ocean bottom. Reverberation returns from these ranges were used to quantify both the strength and spectral character of surface/near-surface reverberation as functions of frequency and environmental conditions. Spectral results over the range of wind speeds (4–17 m/s) have revealed a dominant zero-Doppler component and a weaker-than-expected dependence of spread on both wind speed and frequency. Overall, the results are consistent with sub-surface bubbles as the driver for surface reverberation when whitecaps are present, and will provide data for comparison with theoretical and numerical models, which in turn will give insight into the physical mechanisms responsible for the observed acoustic scattering. [Work supported by SPAWAR (PMW-182) and ONR (Code 4532).]

1:45

1pUW4. Measurements of ocean surface and bottom backscattering strengths in the northwestern Atlantic Ocean. Nolan R. Davis, Joseph Jeffery, and Fred T. Erskine (Acoust. Div., Code 7142, Naval Res. Lab., Washington, DC 20375-5350)

Experimental measurements of ocean surface and bottom back-scattering strengths were carried out in the northwestern Atlantic Ocean during July and August of 1990. The experiment used ship-deployed explosive charges to provide ensonification of the ocean surface and bottom over a range of low frequencies up to 1 kHz. Overall the surface scatter results agreed well with the Ogden-Erskine curves, matching the predicted dependence on wind speed, grazing angle, and frequency. At two sites northeast of the Grand Banks the surface scattering strengths appeared to be dominated by volume scattering due to fish. The bottom scattering strengths were observed to have considerable variation in level (up to 16 dB) between different sites. As a function of grazing angle, most of the bottom scattering strength curves paralleled the Mackenzie curve for grazing angles between 20° and 40°. A moderate frequency dependence of less than 3 dB was observed. Comparisons of the bottom scattering strengths with the Damuth 3.5-kHz echo-character province types yielded no consistent correlations. Comparisons with archival results for the same region yielded general agreement within 6 dB. Comparison of the ship-based and airborne techniques showed that they yielded comparable scattering strengths to within 5 dB.

2:00

1pUW5. High-resolution beam spectra of forward surface scattered signals in shallow water. Yung P. Lee and Charles W. Spofford (Sci. Appl. Intl. Corp., 1710 Goodridge Dr., MS T1-3-5, McLean, VA 22102)

A shallow water experiment was conducted to examine the directionality and spectral spread of cw signals after multiple interactions with the sea surface. Bottomed sources with frequencies between 200 and 300 Hz and two bottomed arrays with apertures up to 250 m were employed allowing source-receiver distances of 6–15 km under a variety of sea conditions and angles of paths with respect to the seas. High-resolution adaptive beamforming techniques were used to suppress grating lobes and compensate for array distortions from linear. Directional beam spectra of the signals were observed illustrating dependences on wind speed and direction with respect to the signal path. The results are interpreted in terms of Bragg scatter from the surface wave.

2:15

1pUW6. Comparison of UMPE/PEREV bistatic reverberation predictions with observations in the ARSRP Natural Lab. Kevin B. Smith (Marine Phys. Lab., Scripps Inst. of Oceanogr., Mail Code 0704, La Jolla, CA 92093-0704), Frederick D. Tappert (University of Miami, Miami, FL), and William S. Hodgkiss (Scripps Inst. of Oceanogr., La Jolla, CA)

During the summer of 1993 acoustic experiment sponsored by the ONR Acoustic Reverberation Special Research Program, bistatic bottom reverberation measurements were made within the ARSRP Natural Laboratory near the mid-Atlantic ridge. Employing a version of the University of Miami Parabolic Equation (UMPE) model [MPL Tech. Memo, 432 (1993)] incorporating Tappert's PE Reverberation (PEREV) model, calculations of monostatic and bistatic reverberation within the Natural Laboratory have been made. The UMPE/PEREV model predicts a strong correlation between areas of high reverberation and high bottom ensonification. Furthermore, the wave scattering strength predicted by the PEREV model is shown to depend on the bottom roughness spectrum evaluated at the Bragg wave number $2k_0 \sin(\phi/2)$ where ϕ is the azimuthal scattering angle. Due to the typical power-law form of the roughness spectrum, this suggests that the wave scattering strength is enhanced in the direction of forward scatter. These predictions will be compared to the observations and the validity of hypotheses will be addressed. [Work supported by ONR, Code 1125OA.]

2:30

1pUW7. Backscatter from gas bubbles in muddy sediments. Frank A. Boyle and Nicholas P. Chotiros (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

Recent experimental evidence suggests that trapped gas bubbles may be dominant in backscatter from sandy sediments. A model based on scattering from a spatial distribution of resonant bubbles was previously used to predict shallow grazing angle backscatter from sandy sediments [F. A. Boyle and N. P. Chotiros, *J. Acoust. Soc. Am.* **93**, 2397 (1993)]. A similar model is developed for predicting backscatter from muddy sites. The sediment gas fraction is treated as a free parameter in fitting model predictions to measured backscattering strengths. Model predictions are compared with experiment. [Work supported by NRL/SSC.]

2:45–3:00 Break

3:00

1pUW8. Azimuthal variation of low-frequency Bragg scattering from the ocean surface. C. S. Hayek, I. W. Schurman, C. A. Boyles, and J. H. Sweeney (The Johns Hopkins Univ., Appl. Phys. Lab., Johns Hopkins Rd., Laurel, MD 20723-6099)

Low-frequency sound, backscattered from an annulus of ocean approximately 9 km wide at a nominal 43-km range, was measured on a 128-hydrophone horizontal line array (HLA) during an experiment in the Gulf of Alaska. Fourier analysis of the backscattered signal permitted separation of the Bragg-shifted returns from the 250-Hz carrier frequency. The Bragg-reflected energy was resolved in azimuth by beamforming the HLA data. Azimuthally varying sea surface backscattering strengths, corresponding to measured wind speeds, were obtained using an exact solution for the backscattered field applied to a two-dimensional Donelan water wave spectrum. Normal mode model predictions using the calculated scattering strengths agreed with the up- and down-shifted Bragg energy measurements in both level and azimuthal dependence. [Work supported by SPAWAR PMW 192.]

3:15

1pUW9. Measurements of bi-static scattering coefficients with frequency dispersive source arrays. Newell O. Booth (NCCOSC, RDT&E DIV, Code 541, San Diego, CA, 92152-6435) and Shelby F. Sullivan, Jr. (Lockheed Aeronaut. Systems Co., San Diego, CA 92152)

A method of measuring bi-static scattering coefficients with frequency dispersive source (FDS) arrays is described and illustrated with experimental data. FDS array pings form a set of narrow beams at different frequencies. Signals from vertical FDS arrays are scattered from boundaries and received on a bi-static receiver. Resolution of the scattering area is achieved from the intersection of the time of arrival ellipse and the conical beam. FDS arrays design principles are summarized and efficient methods of analyzing the data using reverberation models are described.

3:30

1pUW10. Elastic wave scattering by an elliptical inclusion in a fluid-saturated porous medium. Boris Gurevich, Ada P. Sadovnichaja, Sergei L. Lopatnikov (VNII Geosystem, Moscow, Russia), and Sergei A. Shapiro (Geophysical Institute, University of Karlsruhe, Hertzstr. 16, 76187 Karlsruhe, Germany)

The problem of the scattering of an elastic wave by a small (compared to the wavelength of the fast compressional wave) elliptical porous inclusion placed in another fluid-saturated porous medium is studied using the Born approximation. The mechanical behavior of both host and inclusion materials is described by the low-frequency version of Biot's theory. Explicit formulas for the amplitudes of the scattered normal compressional and shear waves and of Biot's slow compressional

wave are obtained. The effectiveness of Biot's slow wave generation depends essentially on the ratio of the wavelength of the slow compressional wave to the inhomogeneity size. For large values of this ratio the results agree with the earlier low-frequency results [J. G. Berryman, *J. Math. Phys.* **26**, 1408-1419 (1985)] derived for a spherical inclusion. In the opposite case new results are obtained. They are used to estimate the effective velocity and attenuation of the normal compressional wave in a porous medium containing randomly distributed inclusions. The frequency dependence of the attenuation is consistent with the results for randomly layered porous materials.

3:45

1pUW11. Higher-order perturbation theory versus the scattering operator expansion method for 3-D surface scattering. John Dubberley (NRL Code 7181, Stennis Space Center, MS 39529-5004), Jorge Novarini (Planning Systems, Inc., Bay St. Louis, MS), and Richard Keiffer (NRL, Stennis Space Center, MS 39529-5004)

To this point little work available in the literature has examined whether three-dimensional scattering events can be successfully modeled by a two-dimensional stimulation. This has mainly been caused by a lack of sufficient computational and modeling power in the past. A contributing reason to the low interest in the validity of this assumption is that there seems to be no compelling reason why the two-dimensional assumption would fail to correctly approximate the three-dimensional situation. However, in the interest of completeness, the ramifications of this assumption must be explored. Here two models are used to provide a preliminary look at this issue. Perturbation theory and the scattering

operator expansion surface scattering methods will be compared to each other over a variety of canonical and "realistic" surfaces. The strengths and weaknesses of each model for this task will be mentioned. Up to third-order implementations of these methods will be used.

4:00

1pUW12. The impulse response of rough two-dimensional surfaces. Richard S. Keiffer (NRL-SSC, Code 7181, Stennis Space Center, MS 39529-5004) and J. C. Novarini (PSI, Long Beach, MS 39560)

Previously, a model for the impulse response of rough two-dimensional (2-D) surfaces, based on an extension of the wedge assemblage method, has been described (R. S. Keiffer *et al.*, Proceedings of the Second IMACS Symposium on Computational Acoustics, Vol. 1, p. 67). In the time that has elapsed since this fully 3-D scattering model's original development, the validity of the WA model, which was questionable for smooth, unwedge-like surfaces has clearly been established for 2-D surfaces [R. S. Keiffer, *J. Acoust. Soc. Am.* (to be published)]. More recently, the extension of the WA method that allows for the application of the technique to fully 2-D surfaces has been rigorously tested in a numerical study involving scattering from thin disks (R. S. Keiffer *et al.*, submitted to *J. Acoust. Soc. Am.*). With the validity of the WA method less in question, and with the increasing availability of benchmark quality solutions (albeit in the frequency domain) for the fully 3-D problem, it seems appropriate at this time to review the 3-D wedge assemblage scattering program (WASP 3D) and to begin a benchmarking exercise. [This work has been supported by ONR.]

MONDAY EVENING, 4 OCTOBER 1993

GRAND BALLROOM D/E, 7:00 TO 9:00 P.M.

Session 1eID

Tutorial on Speech Perception

Judy R. Dubno, Chair

*Department of Otolaryngology and Communicative Sciences, Medical University of South Carolina,
171 Ashley Avenue, Charleston, South Carolina 29425*

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Speech perception. Patricia K. Kuhl (Dept. of Speech and Hear. Sci. and Virginia Merrill Bloedel Hear. Res. Ctr., Univ. of Washington, Seattle, WA 98195)

In this tutorial review new concepts and experimental results on speech perception, particularly its development, will be described. First, an overview is provided of the results of classical experiments investigating speech perception in its "initial state." Next, new results demonstrating the effects of language experience on speech perception will be described, showing that the perception of speech is altered by exposure to language in the first 6 months of life. Finally, perception of auditory-visual illusions in speech perception will be described, including a 90-s demonstration experiment that allows audience members to experience the effect. Auditory-visual effects are discussed within the context of speech perception-production links which have their inception in early infancy. A new theory of the development of speech perception that incorporates these results will be presented.

EQUIPMENT EXHIBIT

See pages xxxi and xxxii for the list of exhibitors.

TUESDAY MORNING, 5 OCTOBER 1993

TERRACE ROOM, 8:00 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography: Acoustical Determination of Polar Ocean Processes II

Subramaniam D. Rajan, Cochair

Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

James H. Miller, Cochair

Department of Electrical and Computer Engineering, Naval Postgraduate School, Monterey, California 93943

Chair's Introduction—8:00

Invited Papers

8:05

2aAO1. The fine scale oceanography of the Arctic Ocean: Doppler acoustic studies. Robert Pinkel and Mark Merrifield (Marine Phys. Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Focus is on fine-scale studies of the upper Arctic Ocean using a variety of Doppler sonars constructed at the Marine Physical Laboratory. From two Spring experiments in the Beaufort Sea (AIWEX, 1985; LEADEX, 1992), considerable insight has been gained into the climatology of the internal wave field. The most energetic constituents of the wave field, near inertial currents, have very different amplitudes and scales than at lower latitudes. The generation of these waves below the ice and lead-covered surface is considered as a possible factor for their unique characteristics in the Arctic. In order to map the current field below an Arctic lead, a sector-scan Doppler sonar has been developed. Using beamforming techniques, the sonar resolves acoustic backscatter and current speed along 28 contiguous beams with an angular resolution of 1.5°. The sonar was deployed at two leads during the LEADEX experiment. Observations were obtained from a 45° vertical fan directed normal to the lead axis. In these leads, convection associated with atmospheric cooling did not modify strongly the background velocity field. However, a clear increase in acoustic backscatter was detected just below the base of the mixed layer, possibly associated with convective processes.

8:30

2aAO2. Tomographic measurements of frontal variability in the Barents Sea. Ching-Sang Chiu (Dept. of Oceanogr., Code OC/Ci, Naval Postgraduate School, Monterey, CA 93943), James H. Miller (Naval Postgraduate School, Monterey, CA 93943), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In August 1992 a coastal ocean tomography experiment was conducted in the Barents Sea over the steep northwestern slope of the Bear Island Trough, about 100 km east of Bear Island. The objective of the experiment was to map and study the oscillations of the Barents Sea Polar Front using acoustic tomography coupled with traditional hydrographic techniques. Because mesoscale ocean variability has shorter spatial and temporal scales in a coastal environment, a vertical receiving array and frequently transmitting tomography sound sources were used to achieve an enhanced system resolution appropriate for coastal monitoring. The vertical array data were processed using plane-wave beamforming to separate ray arrivals in both time and angle. In addition, modal arrivals were separated using broadband modal beamforming techniques. The processed travel time data were then "inverted" using a hybrid ray-mode inverse technique to produce a time series of maps of the polar front. In this presentation, the hybrid ray-mode inverse method and the frontal variability as imaged by this shallow-water tomography system are discussed. [Work supported by ONR 1125AR.]

2aAO3. Observations of acoustic scattering associated with ocean microstructure in the Arctic. Albert J. Plueddemann (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Laurie Padman (Oregon State Univ., Corvallis, OR), Timothy P. Stanton (Naval Postgraduate School, Monterey, CA), Jeffrey T. Sherman, and Robert Pinkel (Scripps Inst. of Oceanogr., La Jolla, CA)

Observations from an Arctic ice camp on the Northwest flank of the Yermak Plateau indicate that under some conditions ocean microstructure may be detectable with "standard" acoustic instrumentation (i.e., acoustic Doppler current profilers) of moderately high frequency (150–300 kHz). The data set, collected during the Cooperative Eastern Arctic Experiment (CEAREX), includes simultaneous observations of kinetic energy dissipation rate, temperature dissipation rate, and acoustic backscatter from both 160- and 300-kHz Doppler profilers. The turbulence levels observed during CEAREX were particularly strong ($\epsilon > 10^{-7} \text{ W kg}^{-1}$) and occurred in well-defined patches. Patches in the mixed layer were not associated with backscattered intensity anomalies. However, intensity anomalies of 2 to 6 dB were found to be coincident in time and space with the patches of strong turbulence in the thermocline. The acoustic intensity anomalies were intermittent, presumably because they were detectable only above a threshold that represented the background particulate scattering level. Theoretical predictions of the acoustic intensity level based on the microstructure measurements are used to support the hypothesis that the enhanced scattering levels are due to temperature microstructure rather than variations in particulate scattering.

9:20

2aAO4. Physical oceanographic and acoustical results from the 1988–89 Greenland Sea tomography experiment. James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Richard Pawlowicz (WHOI), Werner Morawitz, Philip J. Sutton (SIO), and Guoliang Jin (WHOI)

Analyses of the acoustical tomography data taken in the Greenland Sea in 1988–1989, when combined with inputs from other environmental measurements made in the region, have yielded a number of insights into both the physical oceanographic and acoustical properties of that polar sea. The formation of Greenland Sea deep water (GSDW), which is of importance to the global heat engine and climate studies, was imaged tomographically, allowing us to make significant new statements about the mechanisms, rates, and amounts of GSDW formation. Studies of the late arriving acoustic energy have allowed both detailed studies of the evolution of the surface mixed layer as well as a novel acoustical study of the temporal dispersion of acoustic normal modes by that layer in the marginal ice zone. Rough surface scattering processes in ice and open water conditions, as well as ambient noise, have also been studied using the tomography data set.

9:45–10:00 Break

Contributed Papers

10:00

2aAO5. Identification and arrival time estimation of bandpass modes and rays for acoustic tomography in the Barents Sea using a vertical array. James H. Miller, Ching-Sang Chiu (Code EC/Mr, Naval Postgraduate School, Monterey, CA 93943), James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Philip McLaughlin, Christopher W. Miller (Naval Postgraduate School, Monterey, CA 93943), Keith Von Der Heydt, and John Kemp (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In the 1992 Barents Sea Polar Front Experiment, bandpass acoustic tomography signals centered at 224 Hz were received with very high SNR on a 16-element vertical hydrophone array at a range of 35 km from the source. The depth of the ocean varied from 120 m at the source to 280 m at the vertical array. A critical part of travel time tomography is the measurement of the arrival time of individual modes and rays. Identifying and tracking these mode and rays can be extremely difficult using a single hydrophone in coastal waters due to signal interference. A vertical array has the potential to solve these problems. An efficient signal processing scheme is described, written in MATLAB, that simultaneously (1) beamforms for the bandpass modes or rays, (2) compensates for clock drift, sampling skew, and array tilt, and (3) compresses the maximal-length, phase-encoded signal. Data are presented that show that the beamforming along with acoustic modeling provides identification of the arrivals with a high degree of confidence. Errors in measuring the arrival times are estimated to range from 2 ms for low modes to 10 ms for high angle rays making the arrivals useful for inversions. [Work supported by ONR, Code 1125AR.]

10:15

2aAO6. Modal inversion for bottom sound-speed profile in shallow Arctic waters using broadband signals. T. C. Yang (Naval Res. Lab., Washington, DC 20375) and T. Yates (Vector Res. Co., Rockville, MD 20852)

Acoustic inversion of geoacoustic properties in the ocean bottom has been investigated recently using matched-field processing. In the shallow Arctic waters, the acoustic signal may still be dominated by waterborne modes. As a consequence, the matched-field correlation may not be very sensitive to bottom sound-speed variation. It is found that in a water depth of approximately 600 m the matched-field correlation remains very high (> 0.95) even with a mismatch of 50 m/s in bottom sound-speed profile. To improve the sensitivity to bottom mismatch, we include only the high-order modes in the matched-field/mode correlation. This method is investigated using a simulated broadband signal and also using SUS data received on a vertical array. The source is localized using broadband matched-field/mode processing. Short-range data are considered to maximize bottom interacting signals.

10:30

2aAO7. Broadband geoacoustic inversion in shallow Arctic water. John W. Wolf (Naval Res. Lab., Washington, DC 20375)

A broadband technique used successfully to localize acoustic sources and determine water depth and array position at a deep Pacific Ocean

site [P. C. Mignerey, *Oceans '90 Proc.*, 121-124] is used to determine geoacoustic parameters at a shallow water site in the Arctic. The technique minimizes differences between calculated impulse response times and pulse arrival times measured on a vertical array. Initially the method is used to localize the source (SUS) and determine bottom depth and array orientation. The method is then applied to pulse arrivals that penetrate the upper sedimentary layers of the shallow bottom. The inversion provides compressional speed profiles and sediment layer thicknesses of the bottom near the array site. Differences between calculated arrival times and data show a significant decrease, i.e., improvement, when using the geoacoustic profile as determined by the inversion technique. Broadband matched-field processing is used to demonstrate inversion effects on processor performance. The results demonstrate the broadband method to be useful for geoacoustic surveys of shallow water environments.

10:45

2aAO8. Numerical modeling of long-range low-frequency propagation in irregular Arctic waveguide. A. N. Gavrilov, F. I. Kryazhev, and V. M. Kudryashov (N. N. Andreyev Acoust. Inst. and Ocean Acoust. & Inform. Ltd., 38 Vavilov St., Moscow, 1179542 Russia)

An improved model of the Arctic ice cover and the mode coupling approach have been used for numerical calculations of low-frequency sound propagation over the irregular trans-Arctic paths proposed for

acoustic monitoring of Arctic climate change. The ice cover was represented by a rough elastic layer with the roughness statistically determined by its spatial spectrum and correlation between the upper and lower ice surfaces. Horizontal change of the ice statistics along the supposed paths was approximated by a step function in accordance with existing data. The bottom relief and horizontal variation of the typical sound speed profile were approximated by segment-linear functions. In the ice modeling particular attention has been given to the influence of the ice cracks on low-frequency propagation loss. This can be regarded as a shear wave attenuation in the broken ice plate which differs strongly from shear wave absorption in normal ice and mainly depends on a mean size of the ice fields. Calculated acoustic propagation loss over the trans-Arctic paths differ from the empirical to experimental data after a 1000-km distance, with the calculated loss significantly less than the empirical prediction. This is caused by a relative increase of the deep-water mode contribution to the total sound field at such long distances.

11:00-11:15 Break

11:15-12:00

PANEL DISCUSSION:

Panel Moderators: Subramaniam Rajan and James H. Miller

Panel Members: I. Dyer, D. M. Farmer, M. J. Buckingham, J. R. Chamuel, R. Pintel, A. J. Plueddemann, J. F. Lynch

TUESDAY MORNING, 5 OCTOBER 1993

DENVER ROOM, 8:30 TO 11:15 A.M.

Session 2aNS

Noise and Architectural Acoustics: Meeting Room Design: Acoustics or Noise?

Joseph Pope, Cochair

Pope Engineering Company, P. O. Box 236, Newton Centre, Massachusetts 02169

Ewart A. Wetherill, Cochair

Paoletti Associates, 40 Gold Street, San Francisco, California 94133

Invited Papers

8:30

2aNS1. Sound in the meeting room—A listener's perspective. J. Pope (Pope Eng. Co., P. O. Box 236, Newton Centre, MA 02159)

While careful attention is given to the visual appearance of most meeting spaces, the aural appearance often is neglected. Meeting rooms serve a multiplicity of functions in a variety of configurations. Typical of rooms at Acoustical Society meetings, for example, is a lecture-style setup with a presenter at the front of the room addressing an audience of 30-300 listeners. A good room will facilitate the communication of the presenter's message. The listener perceives, comfortably, only what the presenter wants heard. In practice, a number of acoustical defects and distractions may exist. Problem areas include air-handling (HVAC) noise, sound transmission from adjacent spaces, inappropriate reverberation, noise from audio-visual presentation equipment, and audience-generated noise. In addition to bad luck, the causes appear to include design, construction, maintenance, meeting planning, and facility staff issues. The use of a sound reinforcement system can mitigate some problems, while introducing new ones. The intent of this paper is to raise questions for discussion, and to provide an introduction to the session.

8:55

2aNS2. Meeting room design—An architect's perspective. Curtis Worth Fentress (C. W. Fentress J. H. Bradburn and Associates, 421 Broadway, Denver, CO 80203)

C. W. Fentress J. H. Bradburn and Associates was the lead designer for the Colorado Convention Center, Jefferson County Court Building, and for two new airport passenger terminals in Denver and Seoul. From the architect's perspective, there are many considerations in the design of meeting spaces. The author, Principal in Charge of Design, will share his insight into what clients want in design, and what clients think is important to their customers. Acoustics and noise issues are discussed in the context of the overall design project.

9:20

2aNS3. A hotel perspective on meeting room sound. Irv Ash (Carlson Hospitality Group, Inc., 3333 Quebec St., Ste. 114, Denver, CO 80207)

Meeting planners base their decision to stay at a particular hotel, or use a hotel meeting space, on expectations influenced through a travel agent, an advertisement, word of mouth, past experience, or any number of other ways. Regardless of motivation, service and comfort are the primary factors in the decision making. Comfort, like service, is a combination of techniques and ingredients. Discomfort is due to a distortion of these techniques and ingredients. In the hospitality industry, discomfort can often be avoided by getting to know the meeting planner, the needs of the group, and their expectations. Because hotels are generally noisy places, in order to ensure guest comfort it is advantageous when refurbishing or building a hotel to work with a designer who is expert in acoustical treatment.

9:45

2aNS4. Acoustical design of meeting rooms—A moving target. Ewart A. Wetherill (Paoletti Associates, 40 Gold St., San Francisco, CA 94133)

The term *meeting room* encompasses a broad range of facilities, from small, single-purpose conference rooms to very large convention spaces that are capable of being subdivided and of accommodating simultaneous exhibits, banquets, athletics, and performing arts events. Common to all of these uses are the requirements for control of intruding noise and reverberation, optimizing speech intelligibility and overall sound quality, and use of sophisticated audio visual and teleconference systems. These requirements present opportunities for innovative designs in which acoustical flexibility can optimize usability for many diverse functions. Difficulties to be resolved include constraints imposed by competing design requirements, conditions recently mandated by Americans with Disabilities Act (ADA) statutes, severely restricted budgets, and a general lack of understanding of acoustics issues on the part of clients and builders. This paper reviews typical design issues, with examples from recent projects and with particular emphasis on the importance of familiarity with the many acoustical pitfalls inherent in the design and construction process.

10:10

2aNS5. Audio-visual equipment noise in the meeting room—An uninvited participant. Martin Alexander (Briel & Kjaer Instruments, Inc., 63 Passaic Ave., Summit, NJ 07901)

Even meeting spaces that have been designed to have excellent acoustics and low background noise can exhibit noise problems when audio-visual equipment is introduced into the room. Several common A-V equipment types were evaluated with the goal of assessing sound power, tonal content, and directivity. Results of the measurements, and predicted NC and speech interference levels are presented. Concepts for design of quiet equipment are discussed.

Contributed Paper

10:35

2aNS6. An assessment of automotive sound quality. Kathleen K. Hodgdon (Appl. Res. Lab., State Univ., P. O. Box 30, State College, PA 16804)

The intent of this research was to define and validate a statistically sound subjective presentation procedure while gathering additional information on individual preferences for sound quality in the passenger compartment of a six-cylinder car. Samples of noise spectra in the passenger compartments of four six-cylinder vehicles were recorded in both steady and accelerating modes of operation. These noise spectra were then presented in a multiple-comparison listening task to 40 individuals for subjective evaluation and ranking from most to least preferred. A

nonparametric statistical analysis of the data was completed utilizing the Friedman two-way analysis of variance by ranks. The data were analyzed overall as a function of vehicle and as a function of demographic subgroups of the subjects. A retest of 20 subjects was administered approximately 1 month after initial testing to assess the reliability of the procedure. [This research was sponsored by Advanced Powertrain Engineering, Ford Motor Company.]

10:50–11:15

PANEL DISCUSSION

Session 2aPAa

Physical Acoustics: Thermoacoustics

W. Patrick Arnott, Chair

Atmospheric Sciences Center, Desert Research Institute, P.O. Box 60220, Reno, Nevada 89506

Contributed Papers

8:30

2aPAa1. Optimal heat-driven thermoacoustic refrigeration: The beer cooler revisited. W. Patrick Arnott and Anthony A. Atchley (Atmos. Sci. Ctr., Desert Res. Inst., P. O. Box 60220, Reno, NV 89506 and Physics Dept., Naval Postgraduate School, Monterey, CA 93943)

The goals of this research are to design and build an optimally efficient heat-driven thermoacoustic refrigerator to cool a given heat load to a desired temperature. The system uses heat to drive a thermoacoustic prime mover that produces sound for a thermoacoustic refrigerator (as originally investigated by John Wheatley in a device he called a beer cooler on account of the temperatures and heat loads he was striving to obtain). Here, the first step, numerical design, is described. A given system configuration is analyzed using Runge-Kutta integration of three coupled DE's for ambient temperature, acoustic pressure, and specific acoustic impedance (SAI) in the prime mover and refrigerator and using pressure plus SAI transition equations elsewhere. Complex eigenfrequency analysis is used to obtain starting values for a root finding routine for the systems opening acoustic pressure, hot end temperature, heat load, and frequency. Optimization is used to find a system configuration that maximizes overall efficiency. These calculations should shed new insight into optimal heat exchanger lengths. Progress on meeting these goals is reported. The basic set of equations used in the Runge-Kutta integration have intriguing forms and are discussed. Emphasis is placed on investigation of the coupling between prime mover and refrigerator operation. [Work supported by ONR.]

8:45

2aPAa2. Modular thermoacoustic refrigerator. Steven R. Murrell and George Mozurkewich (Ford Motor Co. Res. Lab., Maildrop 3028 SRL, P. O. Box 2053, Dearborn, MI 48121-2053)

A thermoacoustic refrigerator was built to explore scaling to large heat flux. The refrigerator was constructed according to a modular design so that various stack, heat exchanger, and resonator sections are easily interchangeable. The resonator is driven by a commercial 10-in. woofer. Initial tests, using pure helium gas as the working fluid and steel honeycomb (0.8-mm cell) for the stack, pumped 60 W of heat against a 10 °C temperature gradient. Measurements of heat flux and efficiency will be reported as functions of stack structure (e.g., pore size and shape) and will be compared with theoretical predictions.

9:00

2aPAa3. Performance of a thermoacoustic refrigerator with an improved stack geometry. Thomas J. Hofler and Jay A. Adefi (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

The original Hofler thermoacoustic refrigerator is improved by means of a two-segment stack. Each segment of the stack can have a different layer separation that is approximately best for its local acoustic environment and heat load. A few different combinations of stack segments were tested experimentally. The best data indicated an increase in temperature span from 102 to 118 °C, and an increase in COP relative to Carnot of about 40%, compared to the best single-stack element.

9:15

2aPAa4. Thermodynamic performance of a high-power thermoacoustic refrigerator. David K. Perkins, D. Felipe Gaitan, Richard Russell, Ashok Gopinath, L. Todd Morris, James A. Mayfield, and Steven L. Garrett (Phys. Dept. and Space Systems Academic Group-Code Sp/Pk, Naval Postgraduate School, Monterey, CA 93943)

The cooling capacity of a high-power thermoacoustic refrigerator that uses resonant high-amplitude sound in inert gases to pump heat will be measured and discussed. The thermoacoustic engine under study utilizes two thermoacoustic drivers each capable of delivering 60 W of acoustic power at 325 Hz. The engine is designed to deliver 205 W (700 BTU/h) of useful cooling capacity at 4 °C (refrigerator mode) and 117 W (400 BTU/h) at -22 °C (freeze mode). The engine includes two "stacks" with heat exchangers at each end. Using differential thermopiles, the temperature difference across the inlet and outlet of each of the four heat exchangers is combined with the fluid mass flow rate to quantify heat flow. The determination of the acoustic power delivered to the resonator, combined with the heat flows, forms a complete set of thermodynamic measurements which permits accurate determination of the thermoacoustic coefficient-of-performance. These measurements allow optimization of fluid flow rates. [Work supported by NASA—Life Science Division, the NPS Direct Funded Research Program and the Rolex Award for Enterprise.]

9:30

2aPAa5. Effective heat transfer between a thermoacoustic heat exchanger and stack. Thomas J. Hofler (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

Typically, hot and cold copper fin heat exchangers couple heat into and out of the stack. Effective heat transfer means that large amounts of heat are carried across the stack interface with a small temperature difference. Effective heat transfer is also efficient if thermoviscous losses in the heat exchanger are minimized. This is particularly difficult to achieve in engines having high-power density and large gas displacement amplitudes. A simple model for heat exchanger effectiveness and its relationship to fin length and spacing is discussed. Heat exchanger fins that are shorter than the peak gas particle displacement can be effective if the fin spacing is sufficiently small.

9:45

2aPAa6. Experimental heat exchanger performance in a thermoacoustic prime mover. Nelson C. Castro, Thomas J. Hofler, Anthony A. Atchley, and David L. Gardner (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

The effectiveness of heat transfer between the thermoacoustic stack and a typical copper fin heat exchanger is determined by the detailed gas motion and thermal diffusion, and the fin geometry. This effectiveness depends strongly on acoustic amplitude and heat load, as does the ability of the copper to conduct the heat. These and other nonlinear effects contribute to limiting the amplitude of a prime mover. A modular prime

mover experiment has been built so that the limiting amplitude can be measured with a variety of heat exchanger geometries, with the goal of finding the best geometry. The liquid nitrogen temperature experiment uses a heavier gas, such as neon, at a low pressure in order to minimize the heat load, and the temperature defects associated with copper conduction, while allowing large Mach numbers and gas particle displacements. In preliminary measurements, peak pressure amplitudes are in excess of 20% of the mean pressure and the peak-to-peak acoustic displacements are comparable to the stack length. Such large displacements take the study of prime movers into a poorly explored region. [Work supported by the Naval Research Laboratory and the Office of Naval Research.]

10:00

2aPAa7. CFC-free refrigeration: Thermoacoustic technology-design, construction, and evaluation. Joseph Paul Salyards, II (Route 1, Box 302-D, New Market, VA 22844)

Environmental priorities are forcing chlorofluorocarbons (CFC's) out of the marketplace thereby creating a new demand for cooling technologies. The primary objective of this study was to design, construct, and evaluate a CFC-free refrigeration system. A prototype thermoacoustic (TAC) system utilizing high-amplitude sound waves in an inert gas was developed, and several designs were evaluated. The TAC system showed potential as an alternative refrigeration method with the optimum design resulting in a 42.2 °F drop in ambient air temperature. Potential applications exist in refrigeration and air-conditioning; the cooling of satellite-borne sensors; and as either a heat-driven sonar source, cryogenic refrigeration unit, or electric generator.

10:15–10:30 Break

10:30

2aPAa8. Analysis of a thermoacoustics prime mover above onset of self-oscillation. Andrés Larraza and Anthony A. Atchley (Phys. Dept., Code PH/La, Naval Postgraduate School, Monterey, CA 93943)

Nonlinear effects that lead to amplitude saturation in a thermoacoustic prime mover, are described. The evolution of the acoustic amplitude is described by a homogeneous Ginzburg equation of the form $dA/dt = \alpha A - \beta A^3$. The linear term represents the contribution from the power output due to the temperature gradient and viscous and thermal losses. The coefficient α is positive above onset. The cubic term is a consequence of the nonlinear induced vorticity at the boundary layer that originates from irreversible terms. In the approximation considered, the coefficient β is positive and a steady state results from the balance between the linear growth and the nonlinear saturation. The steady-state amplitudes are in qualitative agreement with observations made by Wheatley [Frontiers in Physical Acoustics, Varena (1986)] and Hofer *et al.* [see abstract in this session]. Observations made by Swift [J. Acoust. Soc. Am. **92**, 1551 (1992)] on the dependence of acoustic pressure versus heater power are also in qualitative agreement with the theory. [Work supported by ONR and NPS Direct-Funding Program.]

10:45

2aPAa9. Thermoacoustic termination for a traveling wave tube. John N. Kordomenos (Dept. of Phys., Univ. of Mississippi, University, MS 38677), Anthony A. Atchley (Naval Postgraduate School, Monterey, CA 93943), Richard Raspet, and Henry E. Bass (Univ. of Mississippi, University, MS 38677)

A theory for thermoacoustics of traveling waves has been developed and presented previously by Raspet *et al.* [J. Acoust. Soc. Am. **93**, 2278 (A) (1993)]. In this paper, experimental results are compared with predictions of this theory. The experiment consisted of measuring the pressure reflection coefficient of a thermoacoustic termination located at one end of a 20-m-long, 5-cm-i.d. tube. The termination consisted of a hot heat exchanger, a 5-cm-long, square pore stack, a cold heat ex-

changer, and a cold end. The temperature of the hot heat exchanger was maintained near room temperature. The temperature of the cold end was held close to liquid nitrogen temperature (-77 K). Measurements were made for frequencies ranging from 300 to 1200 Hz, and for different cold end impedances. There is good agreement between the measured and predicted results. [This work was supported by the Office of Naval Research.]

11:00

2aPAa10. Dynamics of a dual-driver acoustic resonator and control system used in a thermoacoustic refrigerator. D. Felipe Gaitan and Steven L. Garrett (Naval Postgraduate School, Monterey, CA 93943)

Until the development of the Thermoacoustic life sciences refrigerator (TALSR) [S. L. Garrett *et al.*, J. Acoust. Soc. Am. **93**, 2364 (1993)], all previous thermoacoustic refrigerators employed only a single electrodynamic driver. Due to the acoustic power requirements for TALSR, a dual-driver, half-wavelength engine was designed. This introduces additional complications for the resonance frequency control and tracking circuitry since each driver constitutes a driven simple harmonic oscillator and the two drivers, both of which have an associated microphone and accelerometer, are coupled by an acoustically resonant system. A computer model of the three strongly interacting resonant systems will be described along with a series of experimental measurements that explored a variety of operating conditions. [Work supported by NASA—Life Sciences Division and the Naval Postgraduate School Direct Funded Research Program.]

11:15

2aPAa11. Cooling effect introduced by acoustic streaming. Junru Wu, Andrew J. Winkler, and Timothy P. O'Neill (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405)

It is believed that a fetal exposure to focused ultrasound during the second and third trimesters of pregnancy represents one of the worst-case situations for thermal effects introduced by ultrasound. This is because (1) the ultrasound beam may pass through a distended bladder that is filled with low attenuating fluid, and (2) heating effects may be most pronounced near well-ossified bone of high attenuation coefficient. It was found in recent *in vitro* experiments that acoustic streaming in low attenuating fluid may dramatically reduce the temperature rise generated by ultrasound at the surface of bone if the ultrasound path contains a low-attenuating-fluid/thin-soft-tissue/bond structure. [Work supported by Hewlett-Packard Co.]

11:30

2aPAa12. Convective heat transfer from a sphere due to acoustic streaming. Ashok Gopinath (Dept. of Phys., Naval Postgraduate School, Monterey, CA 93943) and Anthony F. Mills (UCLA, Los Angeles, CA 90024-1597)

Convective heat transfer from a rigid sphere due to acoustic streaming is discussed for large streaming Reynolds numbers. The work is motivated by a need for the thermal analysis of the experiments on containerless materials processing using acoustic levitation. The acoustic signals used in these experiments give rise to a strong steady fluid motion around levitated alloy samples. The associated convective heat transfer rates are significant enough to give rise to heating/cooling rates in the samples which are crucial in determining their material structure and hence their thermophysical properties. The steady heat transport effect due to the streaming motion is studied with the levitated sample modeled as an isolated solid sphere in a pure plane standing sound field. Strong acoustic signals in the range of 150–170 dB and 1–2 kHz are considered. Analytical and numerical solution techniques are used to obtain a Nusselt number correlation for the heat transfer when the surrounding fluid is a gas with Prandtl number $\sim O(1)$. A simple experiment performed to confirm some of the predictions is found to give very good agreement with the analysis. A more detailed report of the results of this study may be found elsewhere [J. Heat Transfer **115**,

11:45

2aPAa13. Preliminary investigation of high amplitude standing waves with laser Doppler velocimetry. Timothy J. Corrigan and Anthony A. Atchley (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

A previous study of thermoacoustic heat transport phenomena [Atchley *et al.*, *J. Acoust. Soc. Am.* **88**, 251-263 (1990)] reported measurements of the acoustically induced temperature difference ΔT generated across short, poorly thermally conducting plates situated in high

amplitude acoustic standing waves. That study focused on the dependence of ΔT on the position of the plates in the standing wave. Significant discrepancies between the predicted and measured results were observed at high acoustic pressure amplitudes. Moreover, for a given mean gas pressure, there was a threshold acoustic pressure amplitude above which irregularities appeared in the plots of ΔT vs kx . There was evidence that some velocity-dependent effect might be the cause of the discrepancies. An investigation of the acoustic velocity field in high amplitude standing waves has been initiated to determine whether there are measurable irregularities in the velocity field that can account for the observed behavior. Preliminary results of this investigation are reported. [Work supported by the Office of Naval Research.]

TUESDAY MORNING, 5 OCTOBER 1993

GOLD ROOM, 8:30 A.M. TO 12:00 NOON

Session 2aPAB

Physical Acoustics: General Topics

Dale W. Fitting, Chair

Materials Reliability Division, National Institute for Standards and Technology, Boulder, Colorado 80303

Contributed Papers

8:30

2aPAb1. On statistical fluctuations in single scattering by an ensemble of scatterers. George H. Goedecke, Michael De Antonio (Dept. of Phys., NMSU, Las Cruces, NM 88003-0001), and Harry J. Auvermann (Army Res. Lab., Battlefield Environment Directorate, WSMR, NM 88002-5501)

Expressions in terms of the presumed known scattering cross section of the objects are derived for the ensemble average single scattering differential cross section $\bar{\sigma}_N$ and its rms deviation $\Delta\sigma_N$ due to a homogeneous ensemble of N identical objects randomly positioned with no corrections in an observed volume V . For wavelength λ , it is well known that, if $N\lambda^3/V < 1$, there is considerable diffuse scattering, while, if $N\lambda^3/V \gg 1$, the scattering is essentially forward and coherent. Apparently not so well known is the result of $\Delta\sigma_N/\bar{\sigma}_N$ depends on another parameter $\alpha \equiv C(\theta)N(\lambda^3/V)^{4/3}$, when $C(\theta)$ is a numeric of order unity and depends on scattering angle θ . It is shown that, if $\alpha \gg 1$, then $\Delta\sigma_N/\bar{\sigma}_N \propto N^{-1/2}$ and is very small, while if $\alpha < 1$, then $\Delta\sigma_N \approx \bar{\sigma}_N$. Application involving numerical simulation is made to single scattering of acoustic waves by an ensemble of turbules.

8:45

2aPAb2. Locating two uncorrelated sound sources with two vector sound-intensity probes. Robert Hickling (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677) and Alexander P. Morgan (GM Res. Labs., Warren, MI 48090-9055)

The use of vector sound-intensity probes for locating sound sources is a relatively undeveloped area. In a prior presentation [Hickling *et al.*, *J. Acoust. Soc. Am.* **93**, 2357 (A) (1993)] it was shown experimentally that the direction of a single source in water could be detected within ± 2 deg using a probe with four pressure transducers. It was also known that the effect of noise interference from another source could be removed by spectral subtraction. In the present presentation, numerical continuation methods are used to solve a set of nonlinear equations to determine the coordinates and strength of two uncorrelated sound sources from measurements with two vector probes.

9:00

2aPAb3. Nonscattering of sound by sound: Exact second-order transient solutions. Peter J. Westervelt (Dept. of Phys., Brown Univ., Providence, RI 02912)

The bilinear interaction of two primary waves p_1 and p_2 generates a scattered wave p_s satisfying $\square^2 p_s = -\dot{q}_s = -A(p_1\dot{p}_2 + \dot{p}_1 p_2)$, where $A = (\rho_0 c_0^4)^{-1} [2 + \rho_0 c_0^{-2} (d^2 p/d\rho^2)_{\rho_0}]$. Here, $p_1 = p_d + p_m$ was chosen in which $p_d = -\nabla \cdot [(4\pi r) \hat{i} G(t - c_0^{-1} r)]$ and $p_m = -(4\pi c_0 r)^{-1} \dot{G}(t - c_0^{-1} r)$. This combination of dipole and monopole sources located at $r=0$ has a far-field cardioid pattern with a null in the \hat{i} direction. Also chosen was $p_2 = c_0 U(\hat{i} \cdot r + r_0) \sigma [t - c_0^{-1}(\hat{i} \cdot r + r_0)]$, a plane wave originating at $r = -r_0 \hat{i}$ and traveling in the \hat{i} direction. $G(t)$ and $\sigma(t)$ may each have arbitrary time dependence, however, in order to avoid the bilinear interaction of primary waves with their sources, the sources were activated at $t=0$ and terminated at $0 < t < c_0^{-1} r_0$. The function $W = -\frac{1}{2} c_0^2 A (\pi_2 \dot{\pi}_m + \dot{\pi}_2 \pi_m)$ exists in which $\dot{\pi}_2 = p_2$ and $\dot{\pi}_m = p_m$ such that $\square^2 W = \dot{q}_s$, hence $\square^2 (p_s + W) = 0$. Since the far-field solution of this equation is $p_s = -W$ this means that wherever p_2 or p_m and therefore p_1 vanish, so too must p_s vanish. That is, there is no scattering outside the region of interaction. As mentioned earlier [P. J. Westervelt, in *Proceedings of the 13th ISNA*, edited by H. Hobæk (Elsevier Science, London, 1993)] this theory is easily extended to apply to transducers of arbitrary configuration.

9:15

2aPAb4. The FDTD method for ultrasound pulse propagation through a two-dimensional model of the human breast. Charles W. Manry, Jr. and Shira L. Broschat (School of Elec. Eng. & Comput. Sci., Washington State Univ., Pullman, WA 99164-2752)

The incidence of breast cancer is increasing. In 1961 a woman had a 1 in 20 chance of getting breast cancer; today it is 1 in 9. The earlier a tumor is detected, the more likely the survival of the victim. Thus it is critical that high-resolution breast imaging techniques be developed that are capable of detecting small tumors. One modality under consideration is ultrasound imaging. Algorithms developed for ultrasound imaging usually assume that scattering of energy in the breast is weak, allowing simple and efficient image formation algorithms. However,

experiments conducted at the University of Pennsylvania suggest that fat lobes near the breast's surface refract ultrasound energy as it passes through the breast. This strong refraction violates the weak scattering assumption and calls into question the validity of algorithms using this approximation. A higher-order finite-difference time-domain (FDTD) algorithm has been developed that can be used to model the effects of fat lobes on the propagation and scattering of an ultrasound pulse in the human breast. The simple breast model used is two-dimensional with a fat structure similar to that of the Pennsylvania study. Results support the conclusion that refraction in the breast poses a significant challenge to ultrasound imaging. [Work supported by NSF.]

9:30

2aPAb5. Scattering by a layered poroelastic obstacle embedded in a poroelastic host. Raymond Lim (Code 130B, Coastal Systems Sta., Panama City, FL 32407-7001)

A transition-matrix formulation of the field scattered by a three-dimensional, multilayered, poroelastic obstacle embedded in a poroelastic host is obtained by extending Peterson and Ström formulation for scalar waves [B. Peterson and S. Ström, *J. Acoust. Soc. Am.* **57**, 2-13 (1975)]. An exact solution of the vector Biot equations is obtained, obeying all boundary conditions prescribed on all boundaries of the layered obstacle. By numerically implementing the solution for general poroelastic media, scattering with hosts and obstacles made of loss general media is conveniently predicted by choosing material parameters in a limiting sense. Examples are presented to study the scattering dynamics of layered spheres and spherical shells embedded in fluid and poroelastic hosts.

9:45

2aPAb6. Green's functions for wave propagation in range-dependent inhomogeneous media—Some exact analytic solutions. Y. L. Li (US Army Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826-9005)

Three exact analytic solutions of the Helmholtz equation are obtained in closed form for the case of a point source in range-dependent inhomogeneous media. The solutions are obtained by starting with a onefold integral representation for the solution that is derived from the path integral. For a point source in such media above either a soft or hard ground, the exact analytic solutions are also derived. These solutions are particularly useful for testing the validity of approximate or numerical solutions for cases with a range-dependent background medium. They also facilitate construction of more general solutions for problems with scattering in a range-dependent background medium, through the use of integral equation techniques.

10:00

2aPAb7. Performance analysis of transient acoustic wave modeling using higher-order WKBJ asymptotics and symbolical manipulation. Martin D. Verweij (Lab. of Electromagnetic Res., Dept. of Elec. Eng., Delft Univ. of Technol., P. O. Box 5031, 2600 GA Delft, The Netherlands)

The method of modeling transient acoustic waves in continuously layered media using a combination of higher-order WKBJ asymptotic representations and the Cagniard-De Hoop method is addressed. The coefficients of the WKBJ asymptotic representations satisfy a recurrence scheme, which is easily dealt with by applying symbolical manipulation. Moreover, an efficient use of the Cagniard-De Hoop method of inversion is achieved [M. D. Verweij, *J. Acoust. Soc. Am.* **92**, 2223-2238 (1992)]. The performance of this method is discussed on the basis of numerical results for a series of configurations. In particular it is shown how the method performs if transcendental parameter profile functions are replaced by least-square-fit polynomials. From the results it is concluded that the method performs best for low-order polynomial parameter profiles.

10:30

2aPAb8. Computational and experimental investigation of time-shift estimation for compensation of wave-front distortion in ultrasonic applications using scattered signals. Ryoichi Kanda (Med. Eng. Lab., Toshiba Corp., Med. Eng. Ctr., 1385 Shimoishigami Otawara-shi, Tochigi-ken 329-26, Japan), Dong-Lai Liu, and Robert C. Waag (Univ. of Rochester, Rochester, NY 14627)

Time-shift compensation of wave-front distortion that degrades images in medical ultrasound has been studied using computations and measurements. In the computations, wave-front distortion was assumed to arise from a phase screen placed at various distances from the receiving aperture. Intensity patterns were determined from uncompensated data and from data compensated with time-shift estimates made in the aperture. In the experiments, waveforms scattered by a wire target and randomly distributed particles were collected for a water path and for a tissue path. Images were reconstructed from uncompensated data, from data compensated with time-shift estimates made in the receiving aperture, and from data compensated with time-shift estimates made after a backpropagation step in which a waveform similarity factor was maximized. The results show the limitations of time-shift estimation in the aperture for compensation of wave-front distortion produced by a phase screen that is not close to the receiving aperture. The results also show that the addition of a backpropagation step before time-shift compensation can improve the characteristics of the focus.

10:45

2aPAb9. Acoustic band structure in liquids and gases—fcc pattern with spherical balloons. M. S. Kushwaha and P. Halevi (Inst. de Física, Univ. Autónoma de Puebla, Apdo. Post. J-45, Puebla 72750, México)

The first extensive band structure calculations in periodic structures made up of liquids and gases are reported. The specific system studied in the present investigation is spherical air (water) balloons in a periodic pattern in water (air)—the fcc arrangement. Since only longitudinal waves can propagate, the general wave equation is greatly simplified. The pertinent parameters are the densities and the longitudinal-sound velocities. Depending upon the filling fraction a number of band gap(s) extending over the whole first Brillouin zone are found. Within these band gaps vibrations and sound propagation are prohibited. A complete acoustic gap or zero density of states should have important consequences for the realization of noiseless environment and for the localization of longitudinal sound waves. The precise dependence of the band gaps on the filling fraction is also investigated.

11:00

2aPAb10. Photonic band gaps in periodic elastic composites 2-D hexagonal lattices. M. S. Kushwaha and P. Halevi (Inst. de Física, Univ. Autónoma de Puebla, Apdo. Post. J-45, Puebla 72570, México)

The acoustic band structure of periodic elastic composites were calculated using position-dependent density and elastic constant. The specific system investigated is made up of a periodic array of parallel metallic rods of circular cross section whose intersections with a perpendicular plane form a hexagonal lattice. The rods are embedded in a background medium with different elastic constant and density. The transverse polarization was considered—with displacement $u(r,t)$ parallel to the cylinders (and perpendicular to the Bloch wave vectors). The absolute band gaps extending throughout the first Brillouin zone were found in the low-frequency regime. The specific case studied was Cu(Al) cylinders in the Al(Cu) background. A direct comparison of the hexagonal case with the square-lattice pattern (Kushwaha *et al.*, preceding abstract in this session) reveals that the widths of these band gaps are larger in the case of hexagonal lattices. The precise dependence of the gaps on the filling fraction is investigated.

2aPAb11. Coupled wave equations for numerical calculation of acoustical propagation and scattering by atmospheric turbulence. Michael De Antonio, George H. Goedecke (Dept. of Phys., NMSU, Las Cruces, NM 88003-0001), and Harry J. Auvermann (Army Res. Lab., Battlefield Environment Directorate, WSMR, NM 88003-5501)

The usual approximate acoustic wave equations [e.g., A. D. Pierce, *J. Acoust. Soc. Am.* **87**, 2292 (1990)] were developed for ambient conditions (turbulence) in which the scale length a of the spatial variation of temperature and flow velocity is much larger than the acoustical wavelength λ ; this is not the case in many applications of interest. An alternative set of coupled acoustical wave equations, valid for any a/λ , is presented. These equations involve no approximations except the usual neglect of the viscosity terms in the Navier–Stokes equation for the acoustic flow. For quasistatic or “frozen” turbulence, the integral forms of these equations are presented, suitable for application of the digitized Green’s function method [G. H. Goedecke and Sean G. O’Brien, *Appl. Opt.* **27**, 2431 (1988)].

11:30

2aPAb12. First-order acoustical wave equations and scattering by atmospheric turbulence and turbules. George H. Goedecke, Paul M. Pellegrino, Michael De Antonio (Dept. of Phys., NMSU, Las Cruces, NM 88003-0001), and Harry J. Auvermann (Army Res. Lab., WSMR, NM 88002-5501)

The first Born approximation has often been used to predict acoustical scattering by the quasistatic temperature fluctuations and velocities associated with atmospheric turbulence. For this approximation, only terms linear in these disturbances need be retained in the wave equation. Terms in this first-order wave equation involving spatial derivatives of the flow field have often been dropped. It is shown that retaining these terms yields a Born scattering amplitude equal to $\cos(\theta)$ times that

obtained when they are dropped, where θ is the scattering angle. It is also shown that this Born scattering amplitude is identically zero in the forward direction for any solenoidal flow velocity field that goes to zero faster than r^{-3} as $r \rightarrow \infty$. Analytic expressions and numerical results for the first Born differential and total cross sections are obtained for a model localized turbule. The model employs nonuniform rotation about an axis, as modulated by an axisymmetric Gaussian, plus commensurate temperature variation.

11:45

2aPAb13. The study of the influence of boundary layer effect upon the standing wave in an open resonator. Yulia J. Benson (Math. & Mech. Dept., St. Petersburg, State Univ., 2 Bibliotechnaya pl., St. Petersburg 198904, Russia)

A simplest model approach to the problem of the study of nonlinear processes in an open resonator is proposed here. It is based on the consideration of the only act of propagation—reflection of the wave in an open-ended tube. In other words, one can study the incoming wave distortion during wave propagation forward to the closed end (piston) and backward. According to the experimental data the process of oscillation in some cases is mostly defined by these distortions. For example, the experiment [B. Sturtevant, *J. Fluid Mech.* **63**, pt. 1, 97–120 (1974)] predicts that a shock wave should be formed during the propagation considered. The numerical analysis of the standard Burgers’ equation plus the term that was derived by Blackstock [D. T. Blackstock, *J. Acoust. Soc. Am.* **77**, 2050–2053 (1985)] in order to describe an influence of the boundary layer at the walls on plane traveling waves allows one to conclude that an account of this effect permits one to obtain a good agreement with the results predicted by Sturtevant’s experiment. Moreover such a statement can be regarded as an inverse problem, so that it is possible to restore the boundary condition using the experimental data.

TUESDAY MORNING, 5 OCTOBER 1993

VAIL ROOM, 8:25 A.M. TO 12:00 NOON

Session 2aPP

Psychological and Physiological Acoustics: Potpourri

William A. Yost, Chair

Parmlly Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, Illinois 60626

Chair’s Introduction—8:25

Contributed Papers

8:30

2aPP1. Prediction of masked detection thresholds for noise signals centered in a gated noise masker. C. Formby (Dept. of Otolaryngol.—HNS, Johns Hopkins Univ., P.O. Box 1737, Baltimore, MD 21203), M. G. Heinz, and C. E. Luna (Johns Hopkins Univ., Baltimore, MD 21218)

A new detection model can predict masked detection thresholds for a range of signal bandwidths ($W=62$ –6000 Hz) and durations ($T=10$ to 500 ms) for noise signals centered at 2500 Hz in an uncorrelated 500-ms, 6000-Hz-wide masker ($N_0=53$ dB SPL/Hz). The new model effectively takes into account relative cues resulting from temporal and spectral differences between the signal and masker to supplement energy detection. These relative cues are (1) greatest when the durations and spectra of the signal and masker are most disparate and (2) eliminated when the signal and masker are matched temporally and spectrally. The relative timing and spectral shape cues combine independently and

equally to enhance detection by as much as 10–12 dB relative to energy detection alone. The derivation of the new detection model will be explained and predictions from the new model will be compared with experimentally measured masked detection thresholds. The overall agreement between predicted and measured thresholds is excellent ($r=0.95$). [Research supported by NIH.]

8:45

2aPP2. The effects of level and frequency region on the recovery of overshoot. Gary J. Overson and Sid P. Bacon (Psychoacoust. Lab., Dept. of Speech and Hear. Sci., Arizona State Univ., Tempe, AZ 85287-1908)

The term “overshoot” refers to the finding that the threshold for a brief signal can be higher when that signal is presented shortly after masker onset than when it is delayed by a few hundred ms. Overshoot

can be reduced or eliminated by presenting a stimulus (precursor) prior to masker onset. The "recovery of overshoot" can be examined by varying the delay between the offset of the precursor and the onset of the masker. The purpose of experiment 1 was to evaluate this recovery at two (masker and precursor) levels, one relatively high and one relatively low. Overshoot was first measured as a function of masker level in order to choose two fairly disparate levels that produced nearly equivalent amounts of overshoot. Recovery functions were then measured at those levels. In general, there was little difference in the recovery at the two levels. In experiment 2, the precursor was sectioned into two bands, one on either side of the 4.0-kHz signal frequency. Recovery was measured separately for each band, while the other band was presented continuously. There was a tendency for the recovery to depend more upon the band above the signal frequency, suggesting that that region may be most important for overshoot. [Work supported by NIDCD.]

9:00

2aPP3. Effects of amplitude modulation on gap detection in wideband noisebursts. Karen B. Snell (Dept. of Audiol., Rochester Inst. of Technol., Rochester, NY 14618) and Ajit Janardan (Boston Univ., Boston, MA 02215)

Recent studies suggest that fluctuations in the envelopes of narrow-band signals influence the detection of temporal gaps. In this study, the effects of amplitude modulation on gap detection using wideband noisebursts were measured by varying depth of modulation and phase angle. Gap detection thresholds were obtained for four subjects with normal hearing. The digitized noisebursts were 6 kHz in bandwidth and 150 ms in duration. The level of each noiseburst was jittered (70–75 dB SPL) and modulated with a 20-Hz sinusoid. Gaps were placed 100 ms after stimulus onset. Gap detection thresholds were estimated at six modulation depths (0 to 100%) and at one random and three fixed phase angles. A continuous white noise (45 dB SPL) was mixed with the signals. Thresholds were obtained in an adaptive, 2IFC procedure using 100 trial runs. Twenty-four conditions were completed in random order in 2 h. Several replications were completed by subjects. Mean gap detection thresholds increased with depth of modulation and varied with phase angle. Mean gap detection thresholds were smallest (2.5–4.0 ms) for the noisebursts which were least modulated (0–40%). Mean thresholds were largest (20 ms or more) for the noisebursts modulated 100% at a random phase. These results are consistent with recent findings using narrowband signals [B. R. Glasberg and B. C. J. Moore, *Hear. Res.* **64**, 81–92 (1992)].

9:15

2aPP4. Modulation detection interference (MDI) for randomly fluctuating envelopes. L. Mendoza, J. W. Hall, and J. H. Grose (Div. Otolaryngol./Head & Neck Surgery, Univ. North Carolina, Chapel Hill, NC 27599-7070)

Modulation detection interference (MDI) was measured in normal-hearing listeners for stimuli with randomly fluctuating envelopes. The stimuli were generated by modulating pure-tone carriers by dc-shifted low-pass noises (cutoff=10 Hz). The target carrier was 1000 Hz and the interferer carrier 2250 Hz. The target could be presented either alone (baseline condition) or in conjunction with the interferer that was either gated synchronously with the target or was presented continuously. The interferer could be unmodulated, comodulated with the target, or modulated independently from the target. Results indicated substantial MDI in the presence of the interferer both for comodulated and independently modulated envelopes. The differential gating effects will be compared to those for sinusoidal amplitude modulations. [Work supported by the NIDCD R01-DC00418.]

9:30

2aPP5. Some effects of modulator phase on the detection of a signal in a sinusoidally amplitude-modulated tonal complex. Michelle L. Hicks and Sid P. Bacon (Psychoacoust. Lab., Dept. of Speech and Hear. Science, Arizona State Univ., Tempe, AZ 85287-1908)

Subjects were required to detect a 1000-Hz signal in the presence of a masker that consisted of a 1000-Hz (on-frequency) component alone or that component and six flanking components (500, 600, 700, 1300, 1400, and 1500 Hz). The on-frequency component typically was sinusoidally amplitude modulated at 10 Hz and the signal was presented in a dip in the envelope of the on-frequency component or in three successive dips. In experiment 1, the flanking components could lower (comodulation masking release or CMR) or elevate (across-channel masking or ACM) threshold, depending upon whether they were modulated in phase with the on-frequency component, or staggered in phase. CMR was observed for both signal types, although it was smaller for the three-burst signal. ACM was only observed for the three-burst signal. These differences were primarily due to lower thresholds in the presence of the on-frequency component alone for the three-burst signal. In experiment 2, thresholds were measured as a function of the modulator phase of the flanking components. The on-frequency component was unmodulated or 100% amplitude modulated (modulator phase of 0°). For both monotic and dichotic presentations of the flanking components, thresholds increased and then decreased as the modulator phase of the flanking components varied together from 0 to 360°. [Work supported by NIDCD.]

9:45

2aPP6. The effects of cross-spectrum envelope correlation on detection of a signal band: A comodulation detection difference (CDD) experiment. Patrick W. Rappold and John R. Carter (Dept. of Speech Pathol. and Audiol., Univ. of South Alabama, 2000 University Commons, Mobile, AL 36688)

Most CDD and comodulation masking release (CMR) studies employ only two conditions; a *correlated* condition and an *uncorrelated* condition. Temporal envelopes of noise bands are identical in the correlated condition and degree of envelope correlation is the same ($r = 1.0$) across trials within this condition. In the *uncorrelated* condition envelopes of the noise bands are not identical, but degree of envelope correlation fluctuates across trials within this condition. Thus, reported thresholds for correlated conditions represent the average threshold at a single degree of correlation, whereas reported thresholds for uncorrelated conditions represent the average threshold across various degrees of envelope correlation. In this study thresholds are determined for *signal bands* centered at 2000 Hz, in the presence of *cue bands* centered at 1500 Hz, at specific degrees of temporal envelope correlation ($r = 1.0, 0.75, 0.60, 0.50, 0.40, 0.25, 0.0, -0.25, -0.35$). Thresholds at $r = 1.0$ and 0.75 are about 10 dB higher than thresholds at $r = 0.25, 0.0$, and negative degrees of correlation. Thresholds at $r = 0.50$ are about 5 dB less than thresholds at $r = 0.75$ and above, and about 5 dB greater than thresholds at $r = 0.25$ and below. Thresholds at $r = 0.60, 0.50$, and 0.40 are nearly equivalent. These findings suggest that at least three conditions (ranges of correlation) should be employed in CDD, CMR, and other studies investigating the influence of envelope correlation.

10:00–10:15 Break

10:15

2aPP7. Effects of amplitude modulation on profile detection using multiple short signals. Michael A. Smith and Sid P. Bacon (Psychoacoust. Lab., Dept. of Speech and Hear. Sci., Arizona State Univ., Tempe, AZ 85287-1908)

The detection of an increment in a target component is more difficult when the surrounding, nontarget components are sinusoidally amplitude modulated [H. Dai and D. Green, *J. Acoust. Soc. Am.* **90**, 836–845 (1991)]. The purpose of the present study was to determine whether the detrimental effects of amplitude modulation (AM) on profile analysis could be eliminated by providing a relatively "static" profile during the course of AM. The task was to detect an increment in a 1-kHz target component that was centered in a complex containing 21 logarithmically spaced components ranging in frequency from 200 to 5000 Hz. The target component was always unmodulated whereas the nontarget

components were either unmodulated or sinusoidally amplitude modulated at a rate of 10 Hz. The signal was composed of multiple 20-ms presentations, all positioned at successive peaks, valleys, or leading edges of the modulation cycle. As in previous work with longer signals, thresholds generally increased with increasing modulation depth. An attempt to account for the effects of AM in terms of the short-term pedestal resulting from the modulation was only marginally successful. [Work supported by NIH and Sigma Xi.]

10:30

2aPP8. Which ear has the asynchronous signal? Irwin Pollack (Mental Health Res. Inst., Univ. of Michigan, Ann Arbor, MI 48109-0720)

Small differences in the frequency of pure tones are heard as changes in loudness, either when the tones are presented within the same ear (monaural beats) or when presented to the separate ears (binaural beats). With complex tones, comparable differences in fundamental frequency (asynchrony) are heard as changes in roughness or musicality. With two fundamental frequencies presented to each ear, a wide combination of monaural and binaural roughness changes are produced. Under conditions where monaural and binaural roughness changes are clearly detected, the identification of which one of the two ears is asynchronous (one ear asynchronous, the other ear synchronous) is at chance. Substantial interaural differences in intensity are required for the successful uncoupling of the asynchronous ear.

10:45

2aPP9. Auditory apparent motion in the horizontal and median planes. Thomas Z. Strybel (Dept. of Psychol., California State Univ., Long Beach, CA 90840)

The range of stimulus durations and interstimulus onset intervals (ISOIs) that produced the illusion of auditory apparent motion (AAM) were measured when two sound sources were separated in either the horizontal or median planes. Three speaker arrangements were tested, 350° and 10° azimuth at 0° elevation, 0° and 180° azimuth at 0° elevation, and 0° and 20° elevation at 0° azimuth. Subjects were instructed to categorize their perception of the stimulus sequence into one of five categories that were based on the number of sources heard and whether or not motion was perceived. Subjects also indicated the relative location of the leading stimulus. For each speaker arrangement, stimulus durations of 10, 25, and 50 ms and ISOIs between 2 and 70 ms were tested. The plane in which the sources were located did not affect the timing conditions that produced the illusion of motion. AAM was heard for stimulus durations of 25 and 50 ms at ISOI values that were roughly equal to stimulus duration for all of the speaker arrangements tested. However, judgments of the location of the leading source were affected by the arrangement of the speakers. Performance on this task was best when the sources were separated in the horizontal plane.

11:00

2aPP10. Localization performance is enhanced under bimodal conditions of the auditory and visual spatial channels. Brian L. Costantino and David R. Perrott (Psychoacoust. Lab., California State Univ., Los Angeles, CA 90032)

A sequential localization task compared the effects of unimodal versus bimodal spatial acuity in the auditory and visual perceptual systems. The four subjects were tested in a two-alternative forced-choice, three down-one up adaptive paradigm in which two 200-ms signals were presented sequentially with a 1000-ms inter-stimulus interval. Tests were administered with a 620-nm light emitting diode at a luminance

level of 200 ml and a 1.0-kHz high-pass noise at 57 dB (A-weighted). The minimum bimodal angle (MBA) threshold was compared to that of the minimum visible angle (MVA) and minimum audible angle (MAA) thresholds and found to increase performance significantly over the single mode localization tasks, except at 0-deg azimuth that equaled the MVA for most subjects. The implications of these results are discussed.

11:15

2aPP11. A cross-correlation model of common-period grouping. Daniel P. W. Ellis (MIT Media Lab Perceptual Comput. Group, E15-368C, Cambridge, MA 02139)

When extracting and identifying vowels from noisy backgrounds, the human auditory system must identify which frequency regions "belong" to the vowel (as formants), and which are extraneous. This is probably based, at least in part, on the common pitch-period amplitude-modulation of all these regions. Previous computer models of this process have used autocorrelation to identify this cue within each channel of a bandpass array [R. Meddis and M. J. Hewitt, *J. Acoust. Soc. Am.* **91**, 233-245 (1992)]. Autocorrelation, however, is less effective for signals exhibiting rapid frequency modulation or jitter, even though these are strong cues to fusion. A model is proposed that detects comodulation synchrony between frequency bands through a simplified nonlinear approximation to the cross-correlation function. By grouping bands that show a steady modulation skew, pitch-period fluctuation can be used to reinforce grouping rather than weaken it. The physiological plausibility of such a process is also argued.

11:30

2aPP12. Loudness growth of complex stimuli in normal-hearing listeners. Lidia W. Lee (Dept. of Commun. Disord., Northern Illinois Univ., DeKalb, IL 60115-2899) and Larry E. Humes (Indiana Univ., Bloomington, IN 47405)

This paper examined whether an excitation-pattern model of loudness could adequately describe the growth of loudness for complex stimuli presented to normal-hearing listeners in quiet and in noise. The loudness-growth functions were obtained for three synthesized steady-state vowels (/a, i, u/), each with two talkers (male: $F_0=120$ Hz; female: $F_0=200$ Hz), and for several pure tones (125-4000 Hz). All stimuli were presented, at random, from 2 to 90 dB SPL (in 4-dB steps) in quiet and in spectrally shaped noise. Magnitude estimation were used to measure the loudness of each stimulus. These data are used to evaluate the predictions of an excitation-pattern model [E. Zwicker and B. Scharf, *Psychol. Rev.* **72**, 3-26 (1965)] and a simpler modified power-law model [L. Humes and W. Jesteadt, *J. Acoust. Soc. Am.* **85**, 1285-1294 (1989)]. [Work supported, in part, by NIA.]

11:45

2aPP13. Word discrimination in various multitalker noises. Tomasz Letowski, Andrea Tornatore, Jeff Clark, and Barbara MacBeth (Dept. of Commun. Disord., Penn State Univ., 5 Moore Bldg., University Park, PA 16802)

A large number of different multitalker noises (MTNs) is used in research and clinical applications. These noises differ in acoustical, semantic, and linguistic properties and, thus, may have different masking effectiveness. The authors compared acoustical characteristics and masking effectiveness of several MTNs recorded by the authors and by others. Both spectral and temporal differences have been observed among various MTNs. When noises of similar acoustic characteristics have been compared, the gender of the talkers but not the speech material affected masking effectiveness of the MTN.

Session 2aSA

Structural Acoustics and Vibration: Characterization of Viscoelastic Materials I

Bruce Hartmann, Cochair

Naval Surface Warfare Center, 10901 New Hampshire Avenue, Silver Spring, Maryland 20903

Wayne T. Reader, Cochair

Vector Research Company, Inc., 2101 Jefferson Street, Rockville, Maryland 20852

Chair's Introduction—8:55

Invited Papers

9:00

2aSA1. Characterization of viscoelastic materials using the resonant modes of a "free-free" bar. David A. Brown (Phys. Dept., Code PH/Br, Naval Postgraduate School, Monterey, CA 93943)

The technique of using the resonant modes of a "free-free" bar to determine the elastic properties of the bar material will be described. A bar can be electro-dynamically and selectively excited in three independent vibrational modes by mounting small coils of magnet wire on each end of the sample. When the sample ends are placed in a magnetic field, a current through one coil will induce vibrations that can be detected by monitoring the emf produced in a second coil. The elastic shear modulus is determined by measuring the frequency of the torsional mode. The Young's modulus is determined from the frequency of either the longitudinal or flexural mode. The loss tangent, $\tan \delta$, of the material can be obtained by measuring the quality factor of each resonance. Results from a sample using a transfer function technique will be compared with results obtained using a phase-locked loop (PLL). The transfer function technique is convenient, since the voltage across the detection coil is proportional to the velocity of the sample and the current delivered to the excitation coil is proportional to the force delivered to sample. Their ratio is proportional to the impedance of the material. The lock-in method is attractive because it offers a means to continuously monitor the properties of the material.

9:30

2aSA2. Resonance apparatus for characterization of viscoelastic materials. Bruce Hartmann, Gilbert F. Lee, and John D. Lee (Polymer Sci. Group, Naval Surface Warfare Ctr., Silver Spring, MD 20903-5640)

A resonance apparatus for the characterization of viscoelastic materials will be described. Measurements have been made of modulus values as low as 0.1 MPa and as high as 70 GPa and loss factors from 0.003 to 2.5. Material properties are determined as exact (numerical) solutions of the wave equation with no simplifying assumptions. In particular, the loss factor is not calculated from the half-power point and is valid for large values of loss factor. By bonding the sample to the driver, clamping errors common with other instruments are eliminated. For any given measurement temperature, the frequency range of the measurements is about 1.5 decades centered at about 1 kHz. Depending on the sample length and modulus, data can be obtained from 100 Hz to 25 kHz. Once measurements have been made as a function of temperature, the data are shifted to form a master curve and fitted to an analytical model. Illustrative data will be presented for a high-loss and a low-loss polyurethane. [This work was sponsored by the Independent Research Program of the Naval Surface Warfare Center.]

10:00

2aSA3. Accuracy of dynamic modulus measurements made with the dynamic mechanical thermal analyzer. R. J. Deigan and J. J. Dlubac (Code 7451, Ship Acoustics Directorate, Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

The accurate determination of the complex moduli of viscoelastic materials is of great importance for the efficient design of many noise control devices. It is often the case that a design, in conjunction with its acoustic performance requirements, leads to a narrow band of acceptable dynamic modulus for the viscoelastic material. In such cases knowledge of the accuracy and reliability of the measurement apparatus is essential. Dynamic complex shear modulus is measured at Carderock Division Naval Surface Warfare Center (CDNSWC) with the Dynamic Mechanical Thermal Analyzer (DMTA), manufactured by Polymer Laboratories. An attempt to quantify the absolute accuracy of the DMTA measurement has been made through the consideration of errors including the accuracy of the beam solution used to infer the complex modulus, compliance of the sample mounts, measurement noise, temperature control, thermal stress, Poisson effects, and the implementation of time-temperature superposition. With these considerations, an ad hoc error analysis is implemented to quantify absolute error bounds expected for DMTA measurements of a hypothetical material. These results are of direct interest to the ASA Committee on Standards S2WG79, "Characterization of the Dynamic Properties of Viscoelastic Polymers," which has initiated round-robin testing among committee participants. CDNSWC results from the DMTA tests of three round-robin materials will be presented.

10:30

2aSA4. Dynamic mechanical properties of S2WG79 viscoelastic polymer samples and other composite materials. Robert D. Collier (Dept. of Mech. Eng., Tufts Univ., Medford, MA 02115)

S2WG79 candidate materials for standardization have been measured in accordance with procedures recommended by the Working Group. The measurements have been performed with the Tufts viscoanalyzer with a frequency range up to 1 kHz. Results for the candidate materials include complex Young's modulus, shear elastic modulus, and associated loss factors as a function of frequency and temperature. Multi-sample testing provides statistical measures of measurement errors and sample variations. Results are presented in formats for ready comparison with those of other investigators. In addition, results of research on cured epoxy resins and selected dental composites illustrate relationships between measured glass transition temperatures and composite material compositions.

11:00

2aSA5. Servohydraulic applications for vibration damping measurement. Kirk K. Biegler (MTS Systems Corp., 14000 Technology Dr., Eden Prairie, MN 55344-2290)

In recent years, the range of performance has been expanded for servohydraulic test systems utilized for vibration damping measurement. The tests now include: (1) temperature, frequency, and amplitude effects on dynamic characteristics, (2) fatigue effects on dynamic or static properties, (3) resonant frequency determination, (4) stress relaxation after impact, (5) tearing energy analysis, and (6) static deflection measurement. The performance capabilities of servohydraulic test equipment readily available today will be reviewed. Performance capabilities include the range of loading amplitudes, rates, and frequencies and also the automation capabilities that relate to the ease of use and versatility of data analysis currently available. Some of the most recent improvements will be highlighted to show the extended range of tests that servohydraulics can achieve.

Contributed Papers

11:30

2aSA6. Viscoelastic characterization of materials using a dynamic hardness tester. W. Madigosky, R. Fiorito (NSWCWOD, 10901 New Hampshire Ave., Silver Spring, MD 20903-5640), and H. Überall (Catholic University of America, Washington, DC 20064)

The Shore or IRHD hardness test has been widely used for many years and has become one of the standards for characterizing materials in the rubber and plastics industries. The obtained hardness scale values, ASTM or IRHD, are often related to the static elastic properties of rubber-like materials. The classical Hertz equation has also been used to correlate the hardness scale values to the static elastic properties. Recently, progress has been made on the problem of obtaining dynamic viscoelastic properties through the use of a vibrating spherical indenter and modeling the measured input impedance by the radiation impedance of a sphere radiating into a half-space. In the limiting case where the sphere radius is small, such that $ka \ll 1$ for longitudinal waves, it is shown that the measured shear modulus and loss factor are in good agreement with that found using other dynamic methods such as the DMTA or the resonance method. The model is most appropriate for relatively soft materials or very thick hard materials where the thickness of the material can be ignored. The problem of very hard materials and finite thickness where the terminating boundary is important will be discussed.

11:45

2aSA7. Determination of the static moduli of viscoelastic polymers. Wayne T. Reader and David F. Sauter (Vector Res. Co., Inc., 2101 E. Jefferson St., Ste. 700, Rockville, MD 20852)

The characterization of viscoelastic polymers frequently requires determination of the static moduli in addition to determination of the dynamic moduli. A commonly applied method for determining the static Young's or shear modulus is to extrapolate the dynamic modulus in the rubbery regime to vanishing low frequency. A second approach is to relate the hardness, e.g., Shore A, to an estimate of the Young's or shear modulus. As will be briefly described, neither of these approaches appears to provide consistent results. A method is described that exploits the stress/strain hysteresis loop to develop estimates of the static moduli. The method is believed to provide more reliable estimates than obtainable from the previously employed methods. Examples are provided of the method's application to a variety of viscoelastic polymers possessing loss factors varying from about 0.1-1.0, and rubbery moduli varying from 1-10 Mpa. Comparisons are made between the results of the new method and the two previously used ones. [Work supported by ONR 1225AM.]

Session 2aSP

Speech Communication: Applications of Speech Production Research in Disordered Populations

Abeer Alwan, Chair

Department of Electrical Engineering, 66-147E Engr. IV, University of California, 405 Hilgard Avenue, Los Angeles, California 90024

Contributed Papers

8:30

2aSP1. Aerodynamic and acoustic changes following intensive voice therapy for patients with Parkinson's disease: A case study. Christopher Dromey, Lorraine O. Ramig (Recording and Res. Ctr., 1245 Champa St., Denver, CO 80204, and CDSS Dept., Univ. of Colorado, Boulder, CO), and Ronald Scherer (Recording and Res. Ctr., Denver, CO)

A 50-year-old patient with Parkinson's disease received a 4-week course of intensive voice and respiratory therapy aimed at improving functional communication through increased laryngeal adduction. Aerodynamic and acoustic data were gathered on two occasions before treatment, two after the completion of therapy, and at 6- and 12-month follow-up sessions. There were increases in subglottal pressure and laryngeal airway resistance after the treatment. Maximum flow declination rate (MFDR) also increased, as did the sound pressure level of utterances that the subject produced at a "comfortable" effort level. Fundamental frequency variability and conversational intensity levels were higher post therapy. Video-stroboscopic examination before and after treatment showed improvements in adduction and a decreased level of supraglottic hyperfunction. Possible physiological mechanisms are discussed to account for these changes. [Work supported by NIH Grant No. R01 DC01150-03 and NIDCD Grant No. DC00976.]

8:45

2aSP2. The effect of intensive voice therapy on speech and voice deficits in atypical Parkinson's disease. Stefanie M. Countryman (Recording and Res. Ctr., 1245 Champa St., Denver, CO 80204) and Lorraine O. Ramig (Univ. of Colorado, Boulder, CO)

Acoustic data were collected twice pre-treatment, once post-treatment, and 6 months post-treatment on three atypical Parkinson's disease patients following intensive voice therapy. Using a custom-built software program, the following variables were measured: maximum duration of sustained vowel phonation and sound pressure level from sustained vowel phonation, a standard passage, and spontaneous speech. Using the C-Speech application [Milenkovic, J. *Speech Hear. Res.* **30**, 529-538 (1987)], fundamental frequency variability was measured from a standard passage and spontaneous speech. Single-word intelligibility was calculated using naive listeners' perceptions. The patients achieved statistically significant improvement on all variables post-treatment but did not maintain improvement as a group during reading and spontaneous speech by the 6-month data collection session. These results suggest the effectiveness of intensive voice treatment in atypical Parkinson's disease as well as provide insight into the underlying pathophysiology of their speech and voice deficits. Due to the limited length of time post-treatment levels were maintained, a longer period of therapy as well as follow-up therapy sessions are recommended for this population. [Work supported by NIH Grant No. R01 DC01150-03 and NIDCD Grant No. DC00976.]

9:00

2aSP3. Spectral myoelectric analysis of thyroarytenoid activity in normal and spasmodic dysphonia subjects. Rick M. Roark (Dept. of Otolaryngol., New York Medical College, Munger 170, Valhalla, NY 10595), Eric M. Dowling, Ronald D. DeGroat (Univ. of Texas, Dallas, TX), Ben C. Watson, and Steven D. Schaefer (New York Medical College, Valhalla, NY)

Time-synchronous digital recordings of thyroarytenoid (TA) myoelectric activity (via percutaneous hooked-wire electrodes) were obtained for 10 normal control and 11 spasmodic dysphonia subjects during performance of five vocal tract tasks of increasing motoric complexity: (1) quiet breathing, (2) Valsalva maneuver, (3) whispered [i], (4) voiced [i], and (5) *Beep beep went the heap*. Three-dimensional power spectral density functions, $PSD(f,t)$, were determined for a selected segment of the signals, and measures of median frequency, mean frequency, bandwidth, and center frequency of $PSD(f,t)$ were derived for each sample point. Statistical median, mean, standard deviation, maximum, minimum, and mode of the PSD measures were computed to compose feature vectors for each signal. Discriminate analysis and maximum likelihood classification tests were applied to obtain global discrimination measures for the two subject populations as a function of task performance. Results of the analysis are reported that reveal a task effect. Graphical presentations of TA myoelectric and speech acoustic signals with accompanying time-frequency PSD plots are given. Comparisons are made among incidents of spasm and nonspasm, posing questions into physiologic meaning and basis of spectral- versus amplitude-defined laryngeal muscle spasms. [Work supported by NIH Grant No. DC00410 and NSF Grant No. MIP9203296.]

9:15

2aSP4. Quantitative analyses of thyroarytenoid activity in spasmodic dysphonia. Ben C. Watson, Rick Roark, and Steven Schaefer (Dept. of Otolaryngol., Munger, Rm. 170, New York Medical College, Valhalla, NY 10595)

Perceptual ratings of vocal abnormality in spasmodic dysphonia (SD) are confounded by high variability in the severity, quality (strain-strangle versus aspirate), and task sensitivity of symptoms. Perceptual ratings are also relatively distant from the physiologic source of vocal disruption. Investigations at the neuromuscular level are closer to the source of disruption and may be informative. Thyroarytenoid myoelectric activity was recorded while SD and normal control subjects produced multiple tokens of sustained vowel, word repetition, and sentence tasks. Quantitative amplitude measures were submitted to several principal components analyses to develop models of neuromotor abnormality in SD. Analyses yielded quantitative indices of the magnitude and task-sensitivity of thyroarytenoid activity. A between-group model that included one measure across the three tasks achieved 100% discrimination of normal control and SD subjects on one component. The within-group model failed to separate perceptually different subgroups of SD subjects along either component considered separately. Models challenge the validity of the traditional clinical distinction between abductor and adductor SD and the validity of perceptual ratings of vocal

9:30

2aSP5. Comparison between clinician-assisted and computer-assisted methods for obtaining the voice range profile. Darrell Wong, Susan Hensley (Recording and Res. Ctr., Denver Ctr. for the Performing Arts, 1245 Champa St., Denver, CO 80204), Ingo Titze (Recording Res. Ctr., Denver, CO and Univ. of Iowa), Lorraine O. Ramig (Recording Res. Ctr., Denver, CO and Univ. of Colorado, Boulder, CO), and Martin Milder (Univ. of Iowa)

A study was made comparing two methods for obtaining the voice range profile, a clinical tool used to establish a subject's range of intensity and fundamental frequency. One method used a clinician with a pitch pipe to elicit and motivate the subject. The other method used an instructional videotape and a computer. The computer was completely automated, providing visual feedback of the voice range profile, and evaluating each phonation objectively using the criterion of fundamental frequency stability, intensity stability, and duration. The computerized method used a cepstrum algorithm to estimate fundamental frequency, and the intensity was measured by calculating the variance of the signal. The results presented here compare the two methods and discuss the benefits and drawbacks of each approach. Benefits of the computerized method include real time visual feedback of phonatory effort, objective acceptability criterion, and no requirement to phonate at pitches in any particular order. A drawback to the computerized method is its sensitivity to subharmonics. [Work supported by NIH Grant No. DC000-387.]

9:45

2aSP6. A microcomputer-based clinical voice aerodynamic measurement system. Bruce J. Pobjurka (Dept. of Commun. Disord., Univ. of Wisconsin, 1500 Highland Ave., Rm. 461, Madison, WI 53705), Paul H. Milenkovic, and Diane M. Bless (University of Wisconsin, Madison, WI 53705)

The authors' clinical experience with dedicated instruments for voice aerodynamic measurements prompts them to implement these measurements on a microcomputer system. A microcomputer system can be lower in cost, it facilitates archival storage of data, it offers the flexibility of changing what is measured and it offers automated post processing. A system measuring flow, intensity, intraoral pressure, and pitch is implemented on a PC compatible computer, using an Intel 486 microprocessor, running the CSpeech waveform acquisition and analysis software. The salient enhancements to CSPeech used to implement this system are described. The record command was modified to provide visual feedback of voice aerodynamic efficiency, and a pause/resume feature simplifies clinical voice recording procedures. Filter scripts were developed to provide a flexible way to specify analysis procedures and formats for archival storage and retrieval. The filter script is based on the object-oriented concept of specifying both the data and a method for analyzing or archiving the data in the desired form.

10:00-10:30 Break

10:30

2aSP7. Speech input for dysarthric users. Hwa-Ping Chang (Dept. of Mech. Eng. and Res. Lab. of Electron., MIT, Cambridge, MA 02139)

One practical aim of this research is to determine how best to use speech recognition techniques for augmenting the communication abilities of dysarthric speakers. As a first step toward this goal, the following kinds of analyses and tests have been performed on words produced by several dysarthric speakers: a closed-set intelligibility test based on Kent *et al.* [J. Speech Hear. Disord. 54, 482-499 (1989)]; an open intelligibility test; critical listening and transcription; acoustic analysis of selected utterances; and an evaluation of the recognition of words by a commercial speech recognizer. The data from one speaker have been

examined in detail. The analysis and testing have led to several conclusions concerning the control of the articulators for this speaker: production of obstruent consonants was a particular problem (only 30% of syllable-initial obstruents were produced with no error), whereas sonorant consonants were less of a problem (70% correct). Of the obstruent errors, most were voicing errors, but place errors for alveolars (particularly fricatives) were also high, and these consonants were produced inconsistently, as inferred from acoustic analysis and from low scores from the recognizer for words with these consonants. In comparison, vowel errors were less prevalent. Implications for the use of a speech recognizer for augmenting this speaker's communication abilities are discussed.

10:45

2aSP8. Phonatory instability in ALS dysarthria: A case study. Eugene H. Buder (Dept. of Speech and Hear. Sci., Univ. of Washington, JG-15, 1417 NE 42nd St., Seattle, WA 98195) and Edythe A. Strand (Univ. of Washington)

A subgroup of patients with dysarthria resulting from amyotrophic lateral sclerosis (ALS) presents with voice problems characterized primarily by cycle-to-cycle and long-term phonatory instabilities. This paper reports acoustic analyses of the phonation of a 63-yr-old female 9 months post diagnosis of ALS. Perceptual characteristics of voice included inconsistent harshness and cyclic variations in both pitch and loudness. Spectral analysis of fundamental frequency (f_0) and intensity contours revealed periodic instabilities at multiple temporal domains, including 0.4- and 0.8- Hz cycles in both f_0 and intensity ("wow"), 2-Hz cycles in f_0 ("tremor"), and 11-Hz cycles in intensity ("flutter"). Phase relations were noted among these cycles and with other acoustic analysis parameters such as perturbation and spectral energy distribution, indicating that occurrence of short-term and cycle-to-cycle instabilities was modulated by the longer term cycles. The results are discussed with relation to video stroboscopic examinations of this patient, and also in comparison with phonatory characteristics of other patients in this subgroup (females with ALS related voice problems). [Work supported by NIH.]

11:00

2aSP9. Acoustic analysis and synthesis of pathological voice qualities. Christopher Long, Phil Bangayan, and Abeer Alwan (Dept. of Elec. Eng., 66-147E Engr. IV, UCLA, 405 Hilgard Ave., Los Angeles, CA 90024-15940)

An analysis-by-synthesis approach was adopted to classify the acoustic and perceptual features of three pathological voice qualities: breathy, strained, and rough. One hundred and sixty waveforms of the vowel /a/ spoken by female and male subjects with pathological voice qualities were obtained from the VA Hospital in West LA. The temporal and spectral features of the waveforms were studied and the results were used in synthesizing the utterances using the Klatt formant synthesizer. Preliminary results on 30 breathy and strained voices indicate that the perception of "pathological" breathiness is mainly related to: (1) a large open quotient of the glottal waveform (OQ) and (2) the amplitude of aspiration noise (AH) relative to that of voicing (AV) with female voices exhibiting a larger (AH-AV) than male voices. For some voices, it was also necessary to introduce extra poles to the vocal tract transfer function to achieve a better spectral match. Synthesis of strained voices required a lower OQ than that needed for normal voices and, in some cases, amplitude and/or frequency modulation was introduced to achieve a better match in the time domain. The synthetic voices were judged perceptually by clinicians to be of high quality. The results will be discussed in terms of the effects of different vibratory patterns of the vocal folds on the acoustic speech waveform.

11:15

2aSP10. Measurement accuracy of the Glottal Enterprises pneumotach system. Ronald C. Scherer, Christopher Dromey, and

An important instrument for measuring airflows in speech is the pneumotach system provided by Glottal Enterprises. Inverse filtering the measured airflow allows the estimation of glottal volume velocity. The face mask (MSIF-2) has numerous wire-meshed holes placed approximately in the plane of the lips. Airflow from the mouth during speech may come out in a directed stream, creating nonuniform flow through the mask and measured pressures (Micro Switch) that do not reflect the correct flow. Results for cylindrical tubes representing mouth openings indicate that the angle of the flow makes a difference: e.g., for a flow of about 700 cc/s projecting upward versus downward at 45-deg angles, the difference in estimated flow is 100–250 cc/s for a small tube opening (0.2 sq cm). The difference is only about 10 cc/s for a medium size mouth opening of 2.0 sq cm. Also, results indicate that equipment drift is a minor concern, flow estimates are negligibly different for the expected range of face protrusions into the mask for the medium mouth opening, and the effects of temperature change were negligible. Results will be discussed relative to the accuracy of flow measures for particular speech and voice sounds.

11:30

2pSP11. A two-tap pitch predictor for measuring voice aperiodicity noise at high SNR. Mark Leddy (Dept. of Commun. Disord., Univ.

The measurement of aperiodicity noise in the human voice is complicated by the multiple source of this noise. While turbulence noise at the glottis is hypothesized to contribute to aperiodicity noise, it is known that waveform jitter, the cycle-to-cycle fluctuation in pitch period, contributes significantly to the measured aperiodicity noise. In seeking to reduce the influence of jitter on measured noise, it has been observed that a cross-correlation pitch tracker based on a sliding analysis window exhibits the greatest change in measured pitch period when the window crosses a glottal-closure epoch. It was hypothesized that the accuracy of a pitch-predictor based measurement of the noise could be improved if one knew the correct alignment of the analysis window relative to the glottal epoch. In the absence of this knowledge, for each update of a short-term analysis, two alignments of the window were used and that alignment with the least-square pitch predictor error was selected. This leads to a computationally efficient algorithm that does not require explicit determination of glottal epochs. Preliminary trials on subjects with high SNRs indicate a reduced influence of jitter on the observed SNRs.

TUESDAY MORNING, 5 OCTOBER 1993

COLUMBINE ROOM, 7:55 TO 11:30 A.M.

Session 2aUW

Underwater Acoustics: Scattering and Reverberation II

Stanley A. Chin-Bing, Chair

Naval Research Laboratory, Stennis Space Center, Mississippi 39529

Chair's Introduction—7:55

Contributed Papers

8:00

2aUW1. Low-frequency time-dependent scattering from evolving bubble plumes: Theory and numerical results. Robert F. Gragg and Daniel Wurmser (Naval Res. Lab., Code 7144, Washington, DC 20375-5350)

This work advances the capability for modeling the scattering from near-surface bubble plumes beyond the static approximation. Plumes were modeled as evolving structures having finite lifetimes (~ 10 s) with distinct growth and decay stages. The scattering response of such a plume was formulated in terms of the theory of linear time-varying (LTV) filters. Complex scattering amplitudes within the frequency band of interest (50–500 Hz) were computed using either partial wave analysis [for (hemi-) spherical shapes] or boundary integral equation (BIE) methods (for more general shapes). For cw inputs, groups of pseudo-Doppler side peaks were observed in the scattered spectra as a result of the time evolution of the target's internal resonances. Several broadband source waveforms were considered to assess their effectiveness as experimental probes for observing transient near-surface plumes in the field. [Work supported by ONR, Code 4530.]

2aUW2. Abstract withdrawn.

8:15

2aUW3. A sparse boundary-element method for scattering from a rigid body of revolution. Erik M. Rosen (SFA, Inc., 1401 McCormick Dr., Landover, MD 20785), Luise S. Couchman (Naval Res. Lab., Washington, DC 20375), and Francis X. Canning (Rockwell Sci. Ctr., Thousand Oaks, CA 91360)

A sparse boundary-element method is used to calculate the scattered acoustic field of a rigid body of revolution. The $N \times N$ full matrix resulting from the discretization of the Helmholtz integral is transformed to a sparse $N \times N$ matrix using the impedance matrix localization

(IML) method. When applied to integral equations for wave phenomena, the IML method produces a sparse matrix by partitioning the scatterer into several regions, over which the response is represented by basis functions with directional radiation patterns. The number of important physical interactions is reduced to a small number over each region, leaving a large number of matrix elements with relative magnitudes less than 10^{-4} which can be approximated by zero. Used in conjunction with existing BEM codes, the IML method reduces the memory and storage requirements from N^2 to approximately $100N$, while the solution time is $O(N)$ compared to $O(N^3)$ for the full untransformed case. The resulting sparse matrix can be approximately inverted leading to a convergent solution in less than five iterations. Results are given for scattering from a hemispherically endcapped cylinder of $L/a=110$ for $ka=1$ to $ka=17$.

8:30

2aUW4. Multiple scattering effects in scattering from a target in shallow water. Angie Sarkissian (Naval Res. Lab., Washington, DC 20375-5350)

Scattering from a target in a bounded medium involves both propagation effects and target scattering effects as well as the coupling of the two. Previously, in order to simplify the problem, the coupling of the two effects had been neglected by ignoring the contribution of multiple scattering between target and shallow water boundaries. This has been highly desirable because it eliminates the need to evaluate the shallow water Green's function over short ranges. Scattering computations are made to show the importance of including this coupling effect even at relatively low frequencies. It is shown that if the target is closer to the surface than the bottom, then the main contribution of the multiple scattering is due to the interaction of the target with the top surface and this effect may easily be included when the method of superposition is applied.

8:45

2aUW5. Predicting bistatic target scattering from monostatic data. Harry A. Schenck [Code 702(S), RDT&E Div., Naval Command Control and Ocean Surveillance Ctr., San Diego, CA 92152-7520]

Characterizing bistatic scattering from a target for many angles of incidence is expensive and time consuming even in ideal laboratory conditions, and essentially infeasible in an operational environment. Consequently, extending the value of monostatic data by using it to estimate the bistatic target strength is extremely desirable. Previous efforts to do this have severe limitations or restrictions in their applicability. A new technique that combines numerical modeling and experiment will be described. The crux of this method is to rely on modeling to supply the far-field propagator function that predicts the scattering in any direction when the surface values are known, and to determine the surface values by least-squares approximation from knowledge of the monostatic scattering and the principle of reciprocity (which requires the scattering matrix to be symmetric). An example will be shown in which the complete three-dimensional scattering field was estimated for a rigid axisymmetric body given only the monostatic target strength at angles in one plane from 0 to 180 deg. Rules that relate the number of monostatic observations needed to the frequency and the complexity of the target will be described. The extension of this technique to nonrigid scatterers will be discussed.

9:00

2aUW6. Low-frequency backscatter from near-surface bubble plumes. Jeffrey A. Schindall (Dept. of Phys., Univ. of Mississippi, University, MS 38677), Ronald A. Roy, Lawrence A. Crum (Univ. of Washington, Seattle, WA 98105), and William M. Carey (DARPA, Arlington, VA 22203)

Anomalous low-frequency backscatter from the sea surface is gov-

erned by a variety of near-surface effects. Principle among these are microbubble distributions and their subsequent effects on the local sound speed structure. Although a rigorous explanation of these scattering results (including the presence of spikey events often observed in both deep water scattering experiments and sonar trials) would likely require consideration of a host of physical acoustic mechanisms, Occum's razor would dictate that a simple explanation can be both effective and illuminating. This paper examines the role of monopole radiators, the simplest form of radiation, beneath a rough pressure-release surface to explain the presence of low-frequency false targets. It is shown that for certain sea state conditions such false targets can result from transient bubble plumes acting as acoustically compact scatterers. An attempt will also be made to estimate, from the combined probabilities of the presence of white caps and the likelihood that a given bubble plume possesses the requisite volume fraction to produce such an event, the number of false targets to be expected as a function of sea state. [Work supported by ONR and ARPA.]

9:15

2aUW7. Quantitative backscattering coefficients from the numerical scattering chamber. R. A. Stephen and S. A. Swift (Dept. of Geol. and Geophys., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Models of geoacoustic interaction with the seafloor are essential to link the backscattered acoustic field with geological and geophysical descriptions of the seafloor. Beamforming has been implemented into numerical scattering chamber formulation so that backscatter coefficients and functions can be obtained. This process is discussed for a flat, homogeneous seafloor; a single facet on a flat, homogeneous seafloor; and a canonically rough, homogeneous seafloor. This study suggests that representing the backscattered field by a single, angle-dependent coefficient is an oversimplification. Coherence of the scattered field across the beam before stacking and the time spread of the stacked field are significant issues. The numerical stacking chamber computes solutions to the elastic (or anelastic) wave equation by the finite difference method in a two-dimensional Cartesian geometry. The finite difference approach has the following advantages: (a) it includes all shear wave (rigidity) effects in the bottom; (b) it can be applied to pulse beams at low grazing angles; (c) both forward-scatter and backscatter are included; (d) multiple interactions between scatterers are included; (e) arbitrary, range-dependent topography and volume heterogeneity can be treated simultaneously; (f) problems are scaled to wavelengths and periods; and (g) the method considers scattering from structures with length scales on the order of acoustic wavelengths.

9:30

2aUW8. Bottom backscattering strengths derived from shallow-water reverberation measurements. Dale D. Ellis and John R. Preston (SACLANT Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy)

A normal-mode model for calculating surface and bottom reverberation in shallow water has been developed, and found to give good agreement with single-hydrophone data for a fairly flat-bottomed area [D. D. Ellis, SACLANTCEN Report SR-196 (September 1992)]. The model has been extended to include beam patterns and has been compared with broadband data received on a horizontal line array. The model-data agreement is again generally quite good. When the model-data differences as a function of beam angle and time are overlaid on a map of the area, the differences are seen to be primarily due to changes in the bottom bathymetry. This can be used to extract a measure of the bottom backscattering strength versus location.

9:45—10:00 Break

10:00

2aUW9. Reflections and diffractions from facet-wedge constructions: Acoustic experiments and theory. Saimu Li (Dept. of Geol. and Geophys., Weeks Hall, Univ. of Wisconsin, Madison, WI 53706), DeZhang Chu (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and C. S. Clay (University of Wisconsin, Madison, WI 53706)

The higher-order diffractions of transient pressure signal from a combination of hard finite facet wedge of 45 deg and an infinite plane have been studied experimentally and theoretically. Experiments used a spark point source with a separated receiver at the same side of the wedge. The reflected pressure signals were well separated in time domain. Higher-order diffractions from the apex and troughs of the wedge were observed. The studies were concentrated on the reflections from the facets and the diffractions. The experimental geometry is different from Medwin *et al.* [J. Acoust. Soc. Am. **83**, 1794–1803 (1988)]. Biot and Tolstoy's exact wedge solution [J. Acoust. Soc. Am. **29**, 381 (1957)] and extensions were used to compare experimental data and theory.

10:15

2aUW10. Simulation of propagation and scattering in a waveguide with a combined parabolic equation/extended boundary condition model. R. A. Zingarelli (Naval Res. Lab., Acoustics Simulation Sec., Stennis Space Center, MS 39529)

By using the parabolic equation for range-dependent waveguide propagation, the extended boundary condition model for object scattering, and a new method for coupling the two models, basin-scale active acoustic simulations have been performed. The transformations and algorithms, as well as some results, will be presented. [Work supported by ONR.]

10:30

2aUW11. A null field T -matrix perturbation formalism for scattering from a fluid-elastic interface with random surface roughness. Garner C. Bishop (Naval Undersea Warfare Ctr., Div. Newport, Newport, RI 02840)

A null field perturbation formalism for scattering a pressure wave from a fluid-elastic interface with random surface roughness is developed. Helmholtz-Kirchhoff integral equations and the elastic tensor boundary conditions are used to represent the unknown scattered and surface pressure and displacement fields. The null field hypothesis is used to obtain a system of coupled integral equations for the surface fields. Perturbative representations of the scattered and surface fields and of the matrix elements of the Helmholtz-Kirchhoff integral equations are constructed and used to develop equations for the n th-order spectral amplitudes of the unknown fields. It is shown that the n th-order spectral amplitude of a scattered field is coupled to the n th-order spectral amplitudes of all scattered fields and to all lower-order spectral amplitudes of all surface fields. Consequently, the n th-order T -matrix may be calculated recursively and expressed in terms of the zeroth-order off-shell T -matrix elements. The T matrix for the n th-order spectral amplitude of the scattered pressure field in the fluid is calculated, and it is shown that scattering process is mediated by excitation of all possible intermediate surface states created by mode conversion of all lower-order surface field states.

10:45

2aUW12. Modeling and analysis of low-angle ice canopy reverberation in the Arctic. Greg Duckworth, Ted Farrell, Kevin LePage, and Cola Atkinson (Bolt Beranek and Newman, Inc., 70 Fawcett St., Cambridge, MA 02138)

Recent advances in modeling coherent loss in the Arctic have made it possible to attempt a constrained inversion of monostatic Arctic reverberation for the ice scattering matrix. An analysis is presented of

low-frequency active reverberation data collected in the Central Arctic in 1992 where such an inversion yields meaningful results and compare them to theoretical predictions. Historical and measured ice roughness statistics collected in tandem with the acoustic portion of the experiment were used to obtain elastic perturbation theory predictions of scattering loss and backscatter. These losses were integrated into the Kraken normal mode model to yield highly accurate predictions of the coherent signal structure to very long ranges in the Arctic waveguide, as verified by TL measurements. Based on these predictions, a significant portion of the time/frequency structure of the backscattered return received on a 256-element hydrophone array was modeled. To construct a well-conditioned inverse problem, propagation in 10-Hz bands was divided into three distinct groups; surface duct, mid depth and RSR. Modeling of TL and travel time for each band/group made it possible to invert the energy in the backscattered return for the average scattering cross section of the ice cover at the three discrete group grazing angles. Comparison of the derived scattering strengths with the elastic perturbation theory predictions yielded good agreement. [Work supported by ONR.]

11:00

2aUW13. A spectral method for calculation of ocean acoustic normal modes using Chebyshev polynomials. Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., IGPP-0225, UCSD, La Jolla, CA 92093)

A new method employing Chebyshev polynomials to calculate the underwater acoustic normal mode equation was developed. The method is a spectral approach using Chebyshev polynomials as basis functions. This expansion has the advantage of being a particularly efficient and accurate representation of the normal modes (especially of the lower order) since they give an exceedingly good representation of narrow boundary layers such as the sound-speed profile that can undergo rapid changes near the surface. This representation reduces the size of the eigenvalue problem to be solved. Since the CPU time scales with N^3 , where N is the size of the matrix, any size reduction is an advantage computationally. This approach has a significant speed advantage over finite difference methods without sacrificing accuracy. A Chebyshev representation is usually remarkably close to the minimax polynomial that minimizes the maximum error implying high accuracy. Chebyshev polynomials are also useful for the calculation of quantities involving the integral of the mode function, such as modal group velocity, loss, and coupling coefficient matrices in non-adiabatic propagation environments. These quantities can be calculated easily and accurately given their spectral representations without introducing any further numerical error.

11:15

2aUW14. A normal-mode theory of shallow-water reverberation. Renhe Zhang (State Key Lab. of Acoust., Chinese Acad. of Sci., Beijing 100080, People's Republic of China), Guoliang Jin, Dinghua Guan, Wenhua Li, and Xinfang Qiu (Shanghai Acoust. Lab., Chinese Acad. of Sci., Shanghai 200032, People's Republic of China)

Shallow-water reverberation is mainly caused by bottom scattering, and is much more complicated than the deep-water one. Ray theory can be used for calculating short-range reverberation simply by ignoring the water inhomogeneity, but is not suitable for long-range reverberation due to effects of refraction and multipath transmission. Assuming that the scattering sources are homogeneously distributed on the bottom and regarded as the secondary sound sources with certain directivity, by using the WKBZ approximation a normal-mode theory suitable for long-range reverberation is developed. Because the effects of the complex eigenvalues on the mode incident field have been considered, the theory is better than the former ones. The given expressions of reverberation intensity have explicit physical meaning and concise form so that they are easily used for numerical calculation and analytical discussion. Both theory and experiment show that the reverberation in inhomogeneous shallow water has obvious depth dependence and there exists the relationship: $I_R(r, z_1, z_2) = [I_R(r, z_1, z_1) \times I_R(r, z_2, z_2)]^{1/2}$. Some experimental results are given that are consistent with theory.

Session 2pAO

Acoustical Oceanography: Determination of Oceanic Parameters from Scattering in Ocean Waveguides

E. C. Shang, Cochair

CIRES, University of Colorado, NOAA WPL, 325 Broadway, Boulder, Colorado 80303

James F. Lynch, Cochair

Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institute, Woods Hole, Massachusetts 02543

Chair's Introduction—12:55

Invited Papers

1:00

2pAO1. Long vertical arrays in acoustical sounding of the ocean. A. Voronovich¹ (Wave Propagation Lab./NOAA/Environmental Res. Labs., Boulder, CO 80303)

Using long vertical arrays (LVAs) in ocean acoustics enables one to measure an S matrix describing transformations of modes in inhomogeneous waveguides. The S matrix allows one not only to calculate the sound field for any configuration of sources and receivers but also to proceed to the solution of the inverse problem (IP)—retrieving inhomogeneities from acoustical data. The wave-type IP is studied for the 1-D case, which differs significantly from the multidimensional situation. For instance, in the 2-D case, the IP can be solved based only on the submatrix describing forward scattering as a function of frequency. The 2-D irregularities in the acoustic waveguide could be reconstructed as in the 1-D problem with the help of the layer-peeling algorithm via backscattering data. But this approach is of little practical use because backscattering is usually weak. Practical formulation of the deterministic IP in ocean acoustics should exploit some final-dimensional description of the media and use all available data about the S matrix. For example, using phases of diagonal elements measured at only two close frequencies leads to the Munk-Wunsch scheme of modal ocean tomography. In the general case, a numerical-evolution-type algorithm is proposed to solve this nonlinear problem in the spirit of the invariant imbedding approach. [Work supported by NRC and PPSIO of Russia.]

¹On leave from the P. P. Shirshov Institute of Oceanology, Moscow, Russia.

1:25

2pAO2. Renormalization of propagation in layered media with rough interfaces. David H. Berman (Dept. of Phys. and Astron., Univ. of Iowa, Iowa City, IA 52242)

There are now a number of good approximations for plane-wave scattering from rough interfaces separating two half-spaces. This work will describe how these approximations can be used when there are two or more rough interfaces. The technique to be developed is based on matching approximate scattering states, expressed in terms of half-space scattering amplitudes, across source planes. Boundary conditions at interfaces are satisfied by the scattering states. It will be shown how the resulting description can be renormalized so that the mean field is described in terms of effective reflection coefficients at the interfaces, which are distinct from coherent half-space scattering amplitudes. It will also be shown that the method of smoothing gives a first-order approximation to a fully renormalized mean field. Expressions for effective modal attenuations will also be derived.

1:50

2pAO3. Relation between waveguide and nonwaveguide (half-space) scattering from a rough interface. T. F. Gao (Inst. of Acoust., Academia Sinica, Beijing, People's Republic of China) and E. C. Shang (Univ. of Colorado/NOAA/Wave Propagation, Boulder, CO 80303)

A formal solution of the scattering field from a rough interface has been derived. Under the Bass perturbation theory, both waveguide and nonwaveguide (half-space) scattering from a rough interface (the sea bottom) can be calculated analytically. By using an identical definition of "scattering coefficient," a relation between the waveguide scattering (the modal S matrix) M_{ij}^g and the nonwaveguide scattering (conventional plane-wave scattering) coefficient m_i has been found: $M_{ij}^g = (2\pi)^2 m_i / |V(\theta_i)V(\theta_j)|$, where $V(\theta_i)$ and $V(\theta_j)$ represent the reflection coefficients of the plane interface for the modal incident angles θ_i and θ_j , respectively. Consequently, based on this relation, the m_i data measured in the conventional sense (half-space) can be bridged into a shallow water reverberation prediction or vice versa.

2pAO4. Scattering from frontal structures, internal tides, and internal waves during the 1992 Barents Sea polar front experiment. Ching-Sang Chiu, James H. Miller (Naval Postgraduate School, Monterey, CA 93943), Guoliang Jin, Douglas Ray, and James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In August 1992, a coastal ocean tomography experiment was conducted in the Barents Sea over the steep northwestern slope of the Bear Island Trough, about 100 km east of Bear Island. The oceanographic objective of the experiment was to study the dynamics of the polar front and its vicinity using both acoustic tomography and traditional hydrographic techniques. The acoustic objective was to study the effects of the front and other coastal oceanography on the acoustic propagation. These objectives are strongly coupled, in that to effectively perform an inverse for the ocean features, one must first obtain a good understanding of the "forward problem." In this paper, the effect of the strong frontal interface is examined, including its corrugations ("interleaving" features), internal tides, and internal waves on the observed acoustic propagation. Effects on both the modal and ray propagation pictures will be examined, using both theoretical predictions and experimental data.

2:40

2pAO5. Spectral integral formalism of wave propagation in a range-dependent waveguide. Dajun Tang (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543) and Yue-Ping Guo (MIT, Cambridge, MA 02139)

This work is the study of acoustic wave propagation in a range-dependent waveguide, where either the interfaces or the medium parameters are functions of horizontal dimensions. For a range-independent, i.e., horizontally stratified, problem where the wave equation can be solved using separation of variables, the well-established wave-number spectrum formulation has been proved to be a powerful technique. When such a spectrum is known, the modal structure and its continuous component will completely determine the wave behavior in the waveguide. More importantly, knowledge of the spectral information can be used in inversion techniques to estimate environment parameters through acoustic probing. At the ASA Ottawa meeting [Tang and Guo, *J. Acoust. Soc. Am.* **93**, 2284(A) (1993)], Tang and Guo presented the spectrum formalism and an application of the formalism to an ideal wedge. Here the approximate spectral formalism is further developed for more general weakly range-dependent problems. Special attention is paid to the "transition regions" of the waveguide where the discrete part of the spectrum borders the continuous part of the spectrum. Near such regions, different parts of the spectrum interact strongly with each other, resulting in apparent changes of the acoustic fields. Some examples will be discussed to elucidate the advantages of this approach.

3:05–3:15 Break

Contributed Papers

3:15

2pAO6. Matched-field 3-D tomographic inversion for geoacoustic parameters. A. Tolstoy (Naval Res. Lab., Washington, DC 20375)

Matched-field processing has recently been applied to the determination of a wide variety of environmental parameters. However, nearly all efforts have been directed toward determining those parameters along a 2-D swath through the environment. The approach developed for the tomographic 3-D calculation of deep ocean sound-speed profiles [Tolstoy *et al.*, *J. Acoust. Soc. Am.* **89**, 1119–1127 (1991); Tolstoy, *J. Acoust. Soc. Am.* **91**, 781–787 (1992)] has now been adapted for the determination of bottom properties, i.e., sediment thicknesses, sound speeds, and densities. The inversion technique will be discussed and calculations in shallow water environments, where bottom properties are of paramount importance, will be presented.

3:30

2pAO7. A higher-order stochastic model for surface backscatter in high sea states. Kenneth E. Gilbert and Lucy J. Ameling (Appl. Res. Lab. and the Graduate Program in Acoust., Penn State Univ., P. O. Box 30, State College, PA 16804)

A stochastic bubble-layer scattering model based on first-order perturbation theory (plane-wave Born approximation) has recently been developed [K. E. Gilbert, submitted to *J. Acoust. Soc. Am.* (1993)]. In the first-order model, the scattering is treated as a perturbation on the plane-wave solution for the bubble-free water and is written in terms of a "geometric factor" times the horizontal wave-number spectrum of the sound-speed fluctuations in the bubble layer. For low sea states and higher grazing angles, this approach is adequate. For high sea states and low grazing angles, first-order perturbation theory is less accurate and one needs to account for the average bubble density and compute backscatter due to fluctuations about the average. In this paper results are compared for scattering computed with the ordinary plane-wave Born

approximation to that computed with the "distorted-wave" Born approximation (DWBA). It is shown that for high sea states and low grazing angles, distorted waves significantly enhance the backscatter at low frequencies and consequently give better agreement with experiment. The effect can be understood in terms of near-surface upward refraction that creates greater insonification and larger grazing angles near the surface than that obtained with plane waves. Finally, the effect of near-surface upward refraction on rough surface scattering is briefly discussed.

3:45

2pAO8. Experimental investigation of features of low-frequency broadband acoustical signal propagation in shallow water. V. G. Petnikov, A. R. Kozelsky (Inst. of General Phys., Russian Acad. of Sci. and Ocean Acoust. & Information Ltd., 38 Vavilov St., Moscow, 117942, Russia), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In shallow water, the combination of downward refracting sound-speed profiles and strong sound absorption by the bottom limits the range of (tomographic) transmissions and causes one to use lower-frequency broadband signals. In this paper, investigations of low-frequency broadband (25–95 Hz) acoustic signal propagation in a slightly variable shallow waveguide with a negative water column sound-speed gradient are presented. An experiment was carried out in the Barents Sea along a 14-km stationary track. In travel time tomography in shallow water, as in deep water, the resolvability and identifiability of the acoustic arrivals is a necessary condition. In this experiment, seven stable arrival peaks are noted. Over 2 hr, these peaks fluctuate in rms travel time by several milliseconds, which is consistent with the measurement accuracy of the system used. Identification of the arrivals using both mode theory and rays with beam displacement, combined with the ambiguity diagram approach of Munk and Wunsch, has

shown that some of the arrivals are resolvable modes and some are resolvable rays.

4:00

2pAO9. Experimental observations of phase dislocations in shallow water waveguides. A. Yu. Shmelerv, A. A. Migulin (Inst. of General Phys., Russian Acad. of Sci. and Ocean Acoust. & Information Ltd., 38 Vavilov St., Moscow, 117942, Russia), and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Previous work by Petnikov [Akust. Zh. 37, 1212–1215 (1991)] and Shmelerv *et al.* [J. Acoust. Soc. Am. 92, 1003–1007 (1992)] has shown fast phase fluctuations of up to 180° in low-frequency continuous-wave (cw) signals measured during the Coastal Barents Sea Acoustic Test Experiment. These fast fluctuations can be explained as phase dislocations passing through the receiving array [Kravtsov *et al.*, Sov. Phys. Acoust. 35, 156–159 (1989)]. Recent investigations of linearly modulated broadband acoustic signal propagation in the same region have

also shown some features associated with phase dislocations passing by. It was found that the frequency at which a dislocation occurs changes slightly because of ocean variability. This frequency can be considered as a parameter of the acoustic field variability. The results of experimental observations of phase dislocations in several shallow water cw and broadband experiments are summarized. Discussion of these experiments will include suggestions on the use of the dislocation approach in tomographic schemes.

4:15–4:25 Break

4:25–5:00

PANEL DISCUSSION:

Panel Moderators: E. C. Shang, J. F. Lynch

Panel Members: A. Voronovich, D. H. Berman, T. F. Gao, J. H. Miller, C-S. Chiu, D. J. Tang

TUESDAY AFTERNOON, 5 OCTOBER 1993

GOLD ROOM, 1:30 TO 4:10 P.M.

Session 2pEA

Engineering Acoustics: High Power Transducer Materials

Robert Y. Ting, Chair

*Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 568337,
Orlando, Florida 32856-8337*

Chair's Introduction—1:30

Invited Papers

1:35

2pEA1. Future directions in impulsive sound sources. Edward F. Rynne (Naval Command, Control and Ocean Surveillance Ctr., RDT&E Div., 271 Catalina Blvd., San Diego, CA 92152)

While impulsive acoustic sources have long been used by the geophysical community for underwater exploration, sonar applications have been relatively uncommon. Recent work in the area of electric discharge devices has led to both a better understanding of the physics of this class of impulsive sound generator and to new devices, which may in turn lead to an expanded role for such technology in sonar applications—in particular, the observation that the electric arc commonly associated with sparker sound sources represents a wasteful and unnecessary complication. Proper electrode design and control of the electric discharge can eliminate the arc, leaving only a steam bubble, and can thereby enhance the low-frequency performance of such a device. Insight into the performance and potential of such devices is, in part, a result of improved computer modeling capabilities of the nonlinear processes associated with impulsive devices, as well as on the use of high-speed data acquisition in interpreting experimental results. This is especially true for determining the effects of interactions in arrays of bubble sources. Beyond the electric discharge sources, understanding of the coupling of energy from the bubble to the sound field suggests improvements for chemically driven sound sources as well. An update of work on these impulsive devices and the modeling efforts that support them is presented. The performance of some recently developed devices and the potential for future development will be discussed.

2:00

2pEA2. Development of a magnetostrictive sonar transducer using high-temperature superconducting coils. C. H. Joshi, J. P. Voccio (American Superconductor Corp., Two Technology Dr., Westborough, MA 01581), J. F. Lindberg [Naval Undersea Warfare Ctr. (NUWC), New London, CT 06320], and A. E. Clark [Naval Surface Weapons Ctr. (NSWC), Silver Spring, MD 20903]

A low-frequency, dual-piston sonar transducer has been designed, built, and tested. This transducer uses a terbium-dysprosium magnetostrictive rod in combination with high-temperature superconductive (HTS) coils as the driver element. Terbium-dysprosium, when operated at cryogenic temperatures, has the highest known magnetostrictive constant. In this transducer, two HTS coils, driven by a combined dc and ac signal, excite the magnetostrictive element. The resultant motion is transmitted through the vacuum cryostat enclosing the driver assembly to two head masses, which transmit the sound to the surrounding environment. The combination of the high magnetostriction and the absence of electrical dissipation in the HTS coils results in very high overall system efficiency. Modeling of this type of transducer shows overall efficiencies of 80% or more can be obtained. The driver assembly is cooled by a compact Stirling Cycle cryocooler. Performance testing in air and water show

self-resonance occurs at 520 and 430 Hz, respectively. The overall system design, as well as measurements of acoustic output and efficiency, will be presented. [This project was funded by The Naval Undersea Warfare Center, New London, CT, through the Small Business Innovative Research (SBIR) Program.]

2:25

2pEA3. Giant electrostrictive response of polymers. Jerry I. Scheinbeim, Brian A. Newman, and Zhen Yi Ma (Polymer Electroprocessing Lab., College of Eng., Rutgers Univ., P. O. Box 909, Piscataway, NJ 08855-0909)

A new class of electroacoustic materials has been discovered at the Polymer Electroprocessing Laboratory, Rutgers University, based on the enormous electrostrictive response available in certain classes of polymeric materials. Further, many of the properties of these materials, including the dielectric constant and elastic modulus, can be tailored for specific end-use applications. The new material is "created" by applying a very large (> 10 MV/m), dc bias electric field using a superimposed ac driving signal to produce the required acoustic response. The measured thickness response d_T , which is electrostrictive in nature, is proportional to the slope of the strain versus the applied electric field curve and is, therefore, proportional to the applied electric field up to saturation. Extremely high d_T values ($d_T > 10 \text{ \AA/V}$) can be accessed at high enough dc bias fields (> 20 MV/m).

2:50

2pEA4. Development of electrostrictive ceramic compositions for high-power sonar transducers. Robert Y. Ting (U.S. Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

New emphasis on active sonar applications requires the development of compact, high-power sonar projectors. For low-frequency use, one desires to be able to drive the transducer elements at high electric field levels in order to achieve large strains. The conventional lead-zirconate-titanate (PZT) ceramics have been widely used for the design and fabrication of sonar projectors, but these ceramics suffer the shortcomings of strain limitation and nonlinear properties under high drive. In recent years, the Navy has been developing new electrostrictive ceramics of the lead-magnesium-niobate family, because these electrostrictive ceramics can produce much greater strains than the PZTs. In particular, a new class of lead-magnesium-niobate lead-titanate (commonly called PMN-PT) solid solution compositions has been found to offer the combined advantages of high-strain, low-dielectric, and hysteresis losses. The development of these materials is summarized, and the test results from prototype transducers fabricated using the new electrostrictive PMN-PTs are also presented. [Work supported by ONR/T.]

3:15

2pEA5. Materials for thermoacoustic generation of sound. William C. Ward (Los Alamos Natl. Lab., MEE-13, MS J576, Los Alamos, NM 87545) and Thomas B. Gabrielson (NAWC, Warminster, PA 18974)

Thermoacoustic generation of sound underwater represents a technology distinctly different from conventional underwater transduction. By proper design, there is potential for high-power, low-frequency generation of continuous-wave, pulsed, or modulated output. From a materials point of view, there are three major components: the stack, which maintains the temperature gradient responsible for pumping the acoustic oscillations; the working medium, which supports the internal acoustic oscillations; and the heat exchangers, which supply energy from an external source to the stack. Each of these subsystems has critical thermal and acoustical requirements that constrain the selection of materials and the overall source configuration. Thermal conductivity, viscosity, and specific heat replace piezoelectric constants and stress-strain relationships as fundamental properties. In addition, the acoustic performance is modeled more effectively as a waveguide than as a lumped-parameter system. Many of the advantages and disadvantages of this source technology are readily apparent in the context of materials selection and configuration. [Work supported by the Naval Space and Warfare Systems Command, PMW-183.]

Contributed Papers

3:40

2pEA6. Underwater testing of a thermoacoustic sound projector. Steven C. Black (Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

For about the last 10 yr, there has been a growing interest in the thermoacoustic method of transduction. Recently, research has been conducted at Los Alamos National Laboratory on the development of a high-power, low-frequency underwater sound projector based on the concept of a thermoacoustic engine [G. W. Swift, *J. Acoust. Soc. Am.* **84**, 1145 (1988)]. The design of this new transducer consists of a long straight tube deployed vertically. The upper portion of the tube is gas filled and contains the heat engine driver. The lower portion is water filled and acts as an impedance matching section. A small-scale demonstration device has been built and was tested both in air and in water. The demonstration device is approximately 9 m in length and operates in the frequency range of 100 to 200 Hz. In conjunction with experimental testing, modeling of the in-water behavior of this demonstration device was also performed. Results from the modeling and the in-water testing will be presented. [Work sponsored by SPAWAR.]

3:55

2pEA7. Electromechanical properties of new electrostrictive materials for underwater acoustical applications. Kurt M. Rittenmyer and Robert Y. Ting (U.S. Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

Electrostrictive ceramics are known to be capable of generating strains that far exceed those of conventional piezoelectric lead-zirconate-titanate (PZT) ceramics. Several lead-magnesium-niobate (PMN) compositions developed earlier were useful only at higher temperatures than those of interest to the sonar community. A new barium-modified PMN-PT ceramic is described. This material can operate at lower temperatures and has very low losses, making it a prime candidate for high-power sonar transducer applications. Another family of electrostrictive materials capable of generating large strains is lanthanum-modified PZT (or PLZT). These materials exhibit strains up to 0.1% along with extremely high dielectric strength, but their hysteresis losses are higher than those of PMN or PMN-PT ceramics. Since the maximum possible strain output is usually the important parameter in actuator applications, the PLZT family may have potential for new actuator designs. Strain measurements for these new electrostrictive ceramics at high electric fields but different frequency ranges are presented. [Work supported by ONR.]

Session 2pID

Tutorials Committee: Hot Topics in Acoustics

Kenneth J. Plotkin, Chair

Wyle Laboratories, 2001 Jefferson Davis Highway, Suite 701, Arlington, Virginia 22202

A special session in Hot Topics in Acoustics is presented at each meeting of the Society. A member is chosen from each of three or four of the Society's technical committees or specialty groups to present a tutorial talk on topics of current special interest. The talks are intended to help acousticians become familiar with issues and achievements that are not within their own primary field of interest.

Chair's Introduction—3:45

Invited Papers

3:50

2pID1. Hot topics in psychological acoustics. William A. Yost (Parmly Hear. Inst., Loyola Univ. of Chicago, 6525 N. Sheridan Rd., Chicago, IL 60201)

Psychological acoustics, or psychoacoustics, is the study of the relation between sound and the behavior of humans and other animals. Such relations are usually referred to as "hearing." In recent years, the psychoacoustical study of hearing has returned to a major question, "How does an organism determine the sources of sound in its environment, especially when there are a number of simultaneously presented sound sources?" The sensory receptors for hearing are sensitive to the physical properties of the sound field impinging on the organism and are not selective for processing sound sources. The brain must process the neural code of the incoming sound field in order for the organism to determine the sources of the sound that make up the input sound field. The neural code for sound provided by the auditory peripheral nervous system is a temporal-spectral representation of the total sound field. This talk will highlight recent developments in simulating the temporal-spectral code provided by the auditory periphery and advances in the study of properties of this code that might be used by the brain to aid the organism in determining the sources of sounds. [Work supported by NIDCD and AFOSR-Life Sciences.]

4:10

2pID2. Hot topics in physical acoustics. Lawrence A. Crum (Appl. Phys. Lab., 1013 NE 40th Street, Univ. of Washington, Seattle, WA 98105)

Because of the technical area of physical acoustics represents the center of Lindsay's Wheel of Acoustics, this area is often the origin of many research topics that eventually find application in other committees, and even other disciplines. Thus, "hot" topics in physical acoustics can be defined as anything done by physical acousticians, especially if they involve bubbles! However, for the benefit of those from outside this area who want to learn something about this exciting committee, a number of topics that are currently of major interest will be discussed briefly and a few selected topics of particular interest to the presenter will be discussed in more detail. In particular, it will be shown how a few specific topics being developed by physical acousticians have direct relevance to problems of more general interest to our society. For example, advances in the important new area of sonochemistry await progress in the fundamental understanding of sonoluminescence; advances in the new ventures of medical ultrasound, such as extracorporeal shock wave lithotripsy and ultrasonic surgery, await a more complete understanding of acoustic cavitation and nonlinear acoustics; the utilization of containerless processing of exotic materials in space awaits developments in acoustic levitation and an understanding of the surface-oscillation characteristics of liquid drops. These and similar topics will be discussed in some detail.

4:30

2pID3. Hot topics in noise: Sound quality of products. Richard G. Cann (RH Lyon Corp, Cambridge, MA 02138)

The quality of a sound that a product makes is used by the listener in many ways. Broadly, the listener judges from the sound quality the identification of the type of the source. Once identified, the information contained in the sound may be used further. It may be used for practical diagnostic purposes, such as identifying an unanticipated fault or a mode of operation. It may also be used on an aesthetic level to qualify the acceptance of the product. So, in many ways what one hears is used similarly to what one smells or even tastes. Over the years, researchers in psychoacoustics have developed many metrics for sound quality. However, the primary focus has been toward one end of the sound-quality acceptance spectrum—those sounds that annoy or irritate rather than those sounds that gain acceptance or please. The application of sound-quality techniques implies that well-defined specifications for the sound quality must be written so that a manufacturer is clearly able to accept or reject a product. This requirement will be even more acute in the future when many more products will be designed on one continent, built on another, and used on a third.

Session 2pNS

**Noise and Psychological and Physiological Acoustics:
Current Research in Hearing Protection
Device Testing**

Elliott H. Berger, Chair

Cabot Safety Corporation, 7911 Zionsville Road, Indianapolis, Indiana 46268-1657

Invited Papers

1:00

2pNS1. Development of a new hearing protector test standard—Overview of the efforts of ANSI S12/WG11. Elliott H. Berger (Cabot Safety Corp., 7911 Zionsville Rd., Indianapolis, IN 46268-1657)

In 1987, ANSI S12/WG11 (previously designated S12-26) was given an expanded scope and a new title, namely, *Field Effectiveness and Physical Characteristics of Hearing Protectors*. The WG was directed to explore the problems inherent in using optimum-performance laboratory-derived real-ear attenuation data to estimate achievable and/or typical workplace protection, and to identify or develop procedures that could yield improved estimates of field performance. After careful deliberation, the WG determined that a new laboratory-based real-ear attenuation at threshold procedure with explicitly defined subject selection, training supervision, and fitting techniques could potentially provide useful estimates of obtainable field performance. The WG devised a draft protocol and subjected it to a pilot study [Berger *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **88**, S10 (1990)]. Subsequently, the protocol was modified and utilized in a full-scale test of three types of earplugs and one earmuff, by four independent laboratories. The testing was conducted from September 1990 through December 1991. This presentation begins with an overview of the hearing protection rating problem and the rationale for the activities of WG11, as well as an introduction to the remaining invited papers in the session, which focus on the needs for improved hearing protector attenuation data and the results of the interlaboratory study.

1:25

2pNS2. Why users need accurate real-world estimates of hearing protector's ratings. C. Dixon-Ernst (Occupational Health Program, Alcoa, 1501 Alcoa Bldg., Pittsburgh, PA 15219) and A. Behar (Ontario Hydro, Whitby, ON L1N 9E3, Canada)

Accurate values for hearing protectors' attenuations are needed by occupational health professionals who deal with hearing conservation of employees exposed to noise levels in excess of 85 dBA. The problem becomes yet more important in large corporations with extensive use of those devices. Inaccurate NRR values may be the cause of increases in hearing losses, worker's compensation claims, and liabilities. Overestimation of the NRR may result in a decrease in the use of engineering noise controls. Each of these areas will be highlighted and explained.

1:40

2pNS3. Results of the S12 Working Group 11 interlaboratory study to approximate real-world attenuation of hearing protection devices (HPDs). Julia D. Royster (Environmental Noise Consultants, Inc., P.O. Box 30698, Raleigh, NC 27622) and Larry H. Royster (North Carolina State Univ., Raleigh, NC 27607)

Four HPDs, three plugs and one muff, were tested in four labs to identify a protocol that would estimate the real-world attenuation obtained in field studies of workers in hearing conservation programs. Audiometrically experienced subjects who had never worn HPDs regularly used each device, following the manufacturers' instructions without any supplemental experimenter training (subject-fit condition). Subsequently, after the experimenter demonstrated the use of each device, the subject used each HPD again (informed-user-fit condition). Attenuation was measured in two trials with each HPD in each instruction condition. Practice within a condition yielded no significant improvement in achieved attenuation. Experimenter instruction significantly increased the attenuation subjects obtained with earplugs (but not earmuffs). The subject-fit condition approximated the upper quartile of real-world attenuation, and the range of standard deviations of attenuation across labs was no greater than for the informed-user-fit condition, indicating acceptable reproducibility. Testing earmuffs worn with safety glasses only slightly increased the range of standard deviations.

2:05

2pNS4. Hearing protector attenuation from subject-fit methods at the work site and in the laboratory. John R. Franks (Bioacoust. and Occupational Vibration Section, Natl. Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., Mail Stop C-27, Cincinnati, OH 45226-1998) and John G. Casali (Virginia Polytechnic Inst., Blacksburg, VA 24061)

A disparity exists between the advertised and realized attenuation of noise by hearing protective devices. The disparity has been accredited to many factors, including overfitting by experimenters and misuse by workers. This study compared attenuation data for three earplugs and one earmuff obtained from an interlaboratory (IL) study using a subject-fit method with data on the same protectors reported in the literature from work-site (WS) studies where workers fitted the hearing protection and with

manufacturer-reported (MR) data on NRR labels. The IL method required subjects who were audiometrically competent, but inexperienced with hearing protectors, to fit hearing protectors relying solely upon guidance from the manufacturers' written instructions. Results indicated that there were no differences in attenuations between laboratories. Confidence interval testing was performed comparing the IL mean attenuations and variances to WS means and standard deviations and to MR means and standard deviations. Results indicated that the IL data were more consistent with the WS data than with the MR data. In all cases the IL data were significantly less than the MR data. Depending upon protector type, the IL data were equivalent to or greater than the WS data.

2:30

2pNS5. Sample size necessary to provide acceptable reproducibility in laboratory hearing protector attenuation testing. John R. Franks and Carol J. Merry (Bioacoust. and Occupational Vibration Section, Natl. Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., Mail Stop C-27, Cincinnati, OH 45226-1998)

The ANSI standard method for determining real-ear attenuation at threshold requires that no less than ten subjects be used and that each subject be tested no less than three times each with ears occluded and with ears open. Attenuation is calculated as the difference between each paired open and occluded threshold. The grand mean attenuation is calculated for all 30 attenuations. The standard deviation calculation is based on the difference between each attenuation and the grand mean. The ISO standard requires no less than 16 subjects with no less than one calculation of attenuation. From an interlaboratory study using the subject-fit method, it was possible to evaluate the number of subjects and number of repeated measures per subject necessary to reach a desired confidence interval and level of accuracy. The results indicated that the number varies with the type of hearing protector and center frequency of the testing noise. To reach ± 3 dB for an earmuff, as few as 11 subjects with one attenuation test per subject was adequate, while for one of the earplugs more than 75 subjects with two attenuation tests per subject were necessary. Implications of these results on laboratory testing strategies will be discussed.

2:50

2pNS6. Comparison of WG11 study variability to that of other interlaboratory studies. Charles W. Nixon (Armstrong Lab., Crew Systems Directorate, 2610 Seventh St., Wright-Patterson AFB, OH 45433-7901)

The variability of the data measured in the WG11 interlaboratory study was compared with that measured in three other interlaboratory studies conducted in this country, in Europe, and in the Scandinavian nations. All studies utilized a real-ear attenuation at the threshold measurement paradigm and basically followed the procedures described in the current ANSI and ISO hearing protector measurement standards, with one major exception. The WG11 study utilized naive subjects and hearing protector fitting procedures that were relatively free from experimenter influences instead of the trained subjects and the "best fit" procedures required by current measurement standards. A subject fit (*only* hearing protector manufacturer's printed instructions handed to the naive subjects) and an informed user fit (manufacturer's instructions plus experimenter general comments to the naive subjects *only* relative to the instructions) were utilized for the WG11 study measurements. Comparisons indicate that the variability of the data collected under the less rigorous subject and fitting procedures of the WG11 study is at least equivalent to that measured in the other interlaboratory studies.

3:10

2pNS7. A discussion of WG11's anthropometric data with comparison to other available databases. Doug Ohlin (U. S. Army Environmental Hygiene Agency, Bio-Acoust. Div., Aberdeen Proving Ground, MD 21010-5422)

Anthropometric and demographic data were compiled from four participating laboratories on subjects used in WG11's round-robin testing. Age and sex distributions were compared to sizing distributions of the single-flange (V-51R) and EP-100 earplugs. Earplug size was generally found to increase with age, while females were found to be skewed toward the smaller sizes and males toward the larger sizes. Measurements of ear length, ear breadth, pinna protusion, and head height were compared to military standards for anthropometric data.

Contributed Paper

3:25

2pNS8. Measuring the performance of circumaural hearing protectors and communication head sets at low frequencies. J. G. Ryan, E. A. G. Shaw, A. J. Brammer, and G. Zhang (Inst. for Microstructural Sciences, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

An enclosure has been constructed to evaluate the performance of active noise reduction (ANR), circumaural head sets, and hearing protectors, at frequencies below 800 Hz. It consists of a massive (5-kg), flat base plate against which one cup cushion is sealed by spring pressure. The sound pressure within the cup is recorded by a microphone flush-

mounted in the base plate. A 20-cm-diam aluminum tube, 20 cm in length, is attached by annular flanges to the base plate at one end and, at the other, to a 15-cm-diam loudspeaker, which drives the enclosed volume so formed. The excitation of cross modes within the tube is suppressed by a 2.0-cm cylindrical annular layer of felt, in contact with the inner wall of the tube. The performance of the measuring system will be described and results reported for the attenuation of an analog ANR head set. [Work done in collaboration with the Defence and Civil Institute of Environmental Medicine.]

Session 2pPA

Physical Acoustics: Bubbles and Other Topics

Wayne M. Wright, Chair

Physics Department, Kalamazoo College, Kalamazoo, Michigan 49006

Contributed Papers

1:00

2pPA1. Capillary wave turbulence. William Wright, Raffi Budak, and Seth J. Putterman (Phys. Dept., Univ. of California, Los Angeles, CA 90024)

The transition from a discrete spectrum of higher harmonics to a broadband power spectrum has been observed with a wire probe (as regards the frequency spectrum) and imaging of the surface (as regards the wave-number spectrum). By use of laser scattering techniques we hope to obtain an absolute calibration of the surface motion in this very far-off equilibrium state. [Work supported by the U.S. DOE Division of Engineering and Geophysics and by NASA.]

1:15

2pPA2. Absorption of finite-amplitude ultrasound in biomedical tissue. Ping-Wah Li and David T. Blackstock (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

A theoretical investigation of the absorption of finite-amplitude ultrasound in biomedical tissue is reported. The tissue is modeled as a medium having multiple relaxation processes. A generalized Burgers equation, which includes the effects of nonlinearity and relaxation, is derived. Analytical and numerical solutions of this equation for a step shock, a periodic wave, and a temporal Gaussian pulse are presented. The results are used to calculate the finite-amplitude absorption coefficient and temperature rise in the tissue. It is found that nonlinear distortion causes the absorption and the temperature rise to be significantly higher than that for small-signal ultrasound. [Work supported by NIH, ONR, and ARL:UT IR&D.]

1:30

2pPA3. High-frequency (10–25-MHz) wave propagation in argon at pressures up to 10 MPa. C-H. Chiang and L. J. Bond (Ctr. for Acoust., Mech. and Mater., Univ. of Colorado, Campus Box 427, Boulder, CO 80309-0427)

Acoustic wave propagation in gaseous argon has been investigated using a high-power ultrasonic gated-tone-burst transmission system. Results of measurements of compression wave absorption and velocity in argon at frequencies 10 to 25 MHz, at static pressures up to 10 MPa, at temperatures of 21 to 23 °C, and at high power (above critical intensity 16.97 W/m² at 10 MHz and 10 MPa) are presented. For small-amplitude waves at pressures up to 2 MPa, the resulting data are in good agreement with that predicted using ideal gas equations and fundamental thermophysical gas constants. At static pressures higher than 4 MPa and for higher-power settings, distortion of the waveform is observed. Excess attenuation, in addition to classical absorption, is also found at higher power settings. With a 10-MHz fundamental frequency, significant second harmonic generation is observed. The data are compared with existing theories for finite-amplitude waves in fluids. Current theories [F. Dunn *et al.*, IEEE Ultrason. Symp., 527–532 (1981); A. L. Thurais *et al.*, J. Acoust. Soc. Am. **6**, 173–180 (1935)] for harmonic generation and attenuation are shown not to predict the observed ex-

perimental data for high-power, high-frequency waves in argon at high pressure.

1:45

2pPA4. A new explanation for the double resonance of a pure vapor bubble. Yi Mao (Dept. of Phys. and Astron., Univ. of Mississippi, University, MS 38677), Lawrence A. Crum, and Ronald A. Roy (Univ. of Washington, Seattle, WA 98105)

The second resonance frequency $f_0^{(2)}$ of a pure bubble appeared first in R. D. Finch and E. A. Nippiras' theoretical analysis [J. Acoust. Soc. Am. **53**, 1402–1410 (1973)]. However, no one has been able to observe a vapor bubble oscillating at $f_0^{(2)}$. A new analysis based on numerical calculations shows that $f_0^{(2)}$ is physically unstable. By analogizing bubble oscillations with those of a mass-spring system, it is found that the resonance frequency $F(f_d)$ depends on the driving frequency f_d and the real bubble resonance $f_0^{(2)}$ is a solution of $F(f_d) = f_d$. The behavior of $F(f_d)$ near $f_d = f_0^{(2)}$ shows that the bubble tends to shift its oscillation frequency away from $f_0^{(2)}$ (repulsive); whereas $F(f_d)$ near the ordinary resonance $f_0^{(1)}$ is attractive. A theory developed by the authors for directly calculating the resonance frequency and the damping constant without using the mass-spring analogy gives only $f_0^{(1)}$. Therefore, we are forced to the conclusion that $f_0^{(2)}$ results from an improper mass-spring analogy and is not physically observable. [Work supported by ONR.]

2:00

2pPA5. Vapor bubbles revisited. Robert D. Finch (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792)

Mao *et al.* [J. Acoust. Soc. Am. **93**, 2379 (1993)] have revived interest in the subject of the dynamics of vapor bubbles, including the issue of "double" resonances. Some earlier work on this topic will be summarized, and a possible application will be discussed.

2:15

2pPA6. Rectified diffusion and single-bubble sonoluminescence. Ritva Lofstedt, Bradley P. Barber, and Seth Putterman (Phys. Dept., Univ. of California, Los Angeles, CA 90024)

The remarkably low solubility of air in water may be essential to the observation of stable single-bubble sonoluminescence (SL) in this system. In other liquids, the mass exchange during a single cycle appears to be sufficient to quench single-bubble SL. To gain insight into this issue, (1) a theory is developed that unifies mass diffusion and the Rayleigh-Plesset equation, and (2) laser scattering is used as a probe of the time change of bubble parameters due to mass diffusion. [Work supported by the U.S. DOE Division of Advanced Energy Projects and Division of Engineering and Geophysics (Theory); R. L. is an AT&T fellow.]

2:30

2pPA7. Mode-locked sonoluminescence in a sealed system. Robert Hiller and Bradley P. Barber (Phys. Dept., Univ. of California, Los Angeles, CA 90024)

The means whereby synchronous single-bubble sonoluminescence can be achieved in a sealed system will be discussed. This technique is important for the study of gases and liquids other than air and water and also to maintain a constant partial pressure during an extended experimental run. For temporal stability, sonoluminescence and an oscillator can be mode-locked. Also the sonoluminescing resonator can be mode-locked to itself. Measurements that make use of these techniques will be presented. [Work supported by the U.S. DOE Division of Advanced Energy Projects.]

2:45

2pPA8. Comments on the effect of an acoustic field on the shape oscillations of levitated drops and bubbles. Thomas J. Asaki and Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

Acoustic levitation has been used as a method for measuring the frequency and dissipative properties of shape oscillation modes of drops and bubbles [P. L. Marston and R. E. Apfel, *J. Acoust. Soc. Am.* **67**, 27-37 (1980); T. J. Asaki *et al.*, *J. Acoust. Soc. Am.* **93**, 706-713 (1993)]. One of the problems in the interpretation of such measurements is the effect of the levitating field. Some of the relevant issues for slightly deformed drops or bubbles, where the equilibrium shape is spheroidal and R is the radius of the sphere of the same volume, are examined. The interaction of shape oscillations with the external acoustic flow is approximated for a spheroid at the velocity antinode of a standing wave in the long-wavelength limit ($kR \rightarrow 0$) with the result that the effective restoring force for the quadrupole mode is increased. The curvature of the effective potential well is increased. The analysis makes use of the known solution for the potential flow past an incompressible spheroid in the $kR \rightarrow 0$ limit. [Work supported by ONR.]

3:00

2pPA9. Detection and generation of transient acoustic microcavitation at 4.3-MHz center frequency. Qihong Xu, Xiaoyu Zheng, and Robert E. Apfel (Ctr. for Ultrasonics and Sonics, P.O. Box 2159, Yale Station, Yale Univ., New Haven, CT 06520)

To study the potential bioeffects caused by applications of medical diagnostic ultrasound, a highly focused transducer with center frequency at 4.3 MHz was used to generate microcavitation in water seeded with polystyrene particles, and the cavitation was detected by a modified active detection system [R. A. Roy *et al.*, *J. Acoust. Soc. Am.* **87**, 2451-2458 (1990)]. The results show that the cavitation threshold is about 4.0 MPa in water seeded with polystyrene particles of 0.1 μm diameter. The mechanical index corresponding to the cavitation thresh-

old is 2.0, which is in the upper range of diagnostic ultrasound. The results also show that the cavitation depends strongly on the size of the polystyrene particles added in the water. [Work supported by NIH through Grant No. R01 CA39374.]

3:15

2pPA10. Effect of microbubbles on self-similar waterborne shocks. B. Edward McDonald (Naval Res. Lab., Washington, DC 20375)

Similarity solutions developed from the equations of nonlinear acoustics [P. H. Rogers, *J. Acoust. Soc. Am.* **62**, 1412-1419 (1977); B. E. McDonald and J. Ambrosiano, *J. Acoust. Soc. Am.* **84**, 1497-1503 (1988)] have adequately accounted for the range dependence of overpressures in the propagation of underwater spherical shocks, while giving only fair to poor predictions for relaxation time constants at large ranges from the source. The relaxation time discrepancy seems to suggest an anomalously large nonlinearity coefficient at great distances from the source. The effect of minute quantities of entrained air on the shock similarity solution for water is investigated. It is well known that microbubbles can greatly increase the nonlinearity coefficient of water at ambient conditions. At high pressures, however, bubbles may collapse to the point of being dynamically unimportant. The equation of state for water is modified to account for ambient microbubble populations, and the resulting equations solved for self-similar shock profiles. Using air void fraction as a fit parameter, observed relaxation time data are analyzed to yield the levels of entrained air that may have been present in various underwater explosive experiments.

3:30

2pPA11. An electrical-discharge, steam-bubble acoustic source—theory and experiment. Allen H. Olson and Stephen P. Sutton (Naval Command, Control and Ocean Surveillance Ctr., RDT&E Div., Code 711, 271 Catalina Blvd., San Diego, CA 92152-5000)

Electrical energy stored in a high-voltage capacitor bank is rapidly discharged into seawater. Smooth spherical electrodes are used to achieve nearly uniform heating of the seawater in the inner-electrode gap. This allows the water to be superheated, in the liquid state, without electrical breakdown. As the high-pressure, high-temperature liquid expands against the surrounding hydrostatic load, the phase transition to steam occurs. The rapid volume growth of the steam bubble then gives rise to a very broadband, high-frequency acoustic pulse. Experimental data are presented for 1.9-cm-diam electrodes separated by 1.0 cm in 20 Ω cm seawater. The electrical resistance is shown to be a function of the energy delivered to the water. A fully three-dimensional conduction model is presented that takes the electrode geometry and known physical properties of seawater as input. The computation proceeds in constant energy steps, with spatial variation in temperature, electrical conductivity, and electric potential updated at the end of each step. The model accurately predicts the observed electrical resistance and the onset of steam formation.

Session 2pSA

Structural Acoustics and Vibration: Characterization of Viscoelastic Materials II

Wayne T. Reader, Cochair

Vector Research Company, Inc., 2101 East Jefferson Street, Rockville, Maryland 20852

Bruce Hartmann, Cochair

Naval Surface Warfare Center, 10901 New Hampshire Avenue, Silver Spring, Maryland 20903-5000

Chair's Introduction—1:25

Invited Papers

1:30

2pSA1. An overview of fractional-order calculus applied to viscoelasticity. Ronald L. Bagley (Structures Div., Wright Lab., Wright-Patterson AFB, OH 45433)

Over the past 80 yr, an initially obscure branch of calculus has gained some attention as a compact, mathematically convenient description of linear viscoelasticity. Rieman and Liouville developed the classical definitions of fractional-order integration and differentiation. These definitions are fading-memory, linear operators well-suited for describing relaxation, dispersion, and attenuation effects. Early in this century, several authors suggested *ad hoc* fractional-order differentials to describe viscoelastic effects. Scott Blair first suggested the Rieman and Liouville operators in the early 1940s. Several investigators have subsequently echoed Scott Blair's suggestion. More recently, the fractional-order calculus viscoelasticity formulation has been expanded to include dynamics and finite-element formulations. Most recently, the formulation has led to the development of the thermorheologically complex model, a generalization of the thermorheologically simple material model, and the model has been related to molecular bond energy decay and fractals. Other recently developed engineering applications of fractional calculus include corrosion, aeroelasticity, and feedback control.

2:00

2pSA2. A new shift factor for the characterization of the dynamic properties of viscoelastic materials. Ahid D. Nashif (Roush Anatrol, Inc., 10895 Indeco Dr., Cincinnati, OH 45241)

During the course of characterizing the dynamic properties of a variety of viscoelastic materials, over the last 10 yr, the Arrhenius equation was found to be valid for describing the shift factor over the glassy, transition, and rubbery regions of the material. Based on this observation, the work described was initiated to define the activation temperature in terms of the dynamic behavior of viscoelastic materials. It will be shown, both empirically and analytically, that the activation temperature is a unique function of the dynamic behavior of viscoelastic materials, which can be accurately determined once the properties of the material have been measured for any frequency but over a wide temperature range. Two advantages exist for using this unique expression to describe the shift factor instead of the previously used ones. The first is that it eliminates the cumbersome and sometimes questionable ways of curve-fitting the data. The second, which is more important, is that by using this unique function to reduce the data results in a smooth set of points without discontinuities; otherwise, the measured data are questionable, and their validity should be investigated.

2:30

2pSA3. Standard viscoelastic material. R. E. Wetton, J. C. Duncan, and G. M. Foster (Polymer Labs. Ltd., The Technology Ctr., Epinal Way, Loughborough LE11 0QE, UK)

Whereas the moduli of elastic solids and the viscosities of simple molecular liquids are well-defined, such that many can be used as standards, the same is not true of viscoelastic materials. The properties of the latter are by definition both frequency and temperature dependent. In this paper some candidate materials and the results obtained over a wide (-100 to $+100$ °C) temperature range and a fairly wide measurement frequency range (0.01–100 Hz) are considered. In order to extend this frequency range, particularly into the kHz range, the time-temperature superposition ideas of Williams, Landel, and Ferry are

used. This necessitates the material conforming to this treatment by virtue of possessing a temperature-independent relaxation spectrum, to a good approximation. In order to have relatively high damping at ambient, the polymer has to have a T_g around 0 °C, and, to be thermorheologically simple, it has to be predominantly single phase. The best candidate materials have been found to be random copolymers of butadiene and acrylonitrile.

Contributed Papers

3:00

2pSA4. Characterization of candidate standard polymers. Wayne T. Reader, N. Scott Emery, and Lois S. Pena (Vector Res. Co., Inc., 2101 E. Jefferson St., Ste. 700, Rockville, MD 20852)

Investigators actively engaged in the measurement of the moduli and loss factors of viscoelastic polymers are well aware of the diversity of results obtained by different measurement techniques—even though the specimens are fabricated to be identical. To alleviate this perennial problem, the ASA has sponsored working group S2WG79 to identify several polymers for use in calibration of the multitude of apparatuses being used to characterize dynamic moduli and loss factors. Criteria used to select candidate polymers include ready availability, property stability, bondability to mounting blocks, versatility to cover wide frequency and temperature ranges, manufacturing repeatability, and, most important, rheological simplicity. Three candidate polymers have been proposed and distributed for initial evaluation. Evaluations of how these candidates performed on a Polymer Laboratories DMTA will be presented, and their suitability for standard materials as determined by the selection criteria will be addressed.

3:15

2pSA5. Efficiency of viscoelastic materials in the vibration reduction of thin rectangular plates. R. Lapointe and L. Cheng (Mech. Eng. Dept., Université Laval, Québec G1K 7P4, Canada)

The problem of reducing structural vibration with viscoelastic materials while adding the least extra weight possible has always been of great interest, especially in the aeronautic field. To study this, the system consisting of a thin rectangular plate covered by (unconstrained) sections of viscoelastic material that are also rectangular will be used. The minimization of the vibration level of the plate when subject to external forces using a determined weight of coverage will then be treated, with special attention given to the frequency- and temperature-dependent characteristics of the viscoelastic materials. The mathematical model of the system is first developed following the Love–Kirchoff plate theory and a polynomial Rayleigh–Ritz approximation. The search for the minimal level, using the simulated annealing technique, is then initiated. Numerical results will be presented and opportunities for using these results for more complex systems will be discussed.

3:30

2pSA6. Measurement of changes in the bulk material properties of solid rocket motor propellant. Rosario Nici (Dept. of Astronautics, USAF Acad., Colorado Springs, CO 80840) and Leonard J. Bond (Univ. of Colorado, Boulder, CO 80309-0427)

Solid rocket motor propellant's bulk viscoelastic material properties change with age and service history. Safety considerations preclude the use of propellant after an estimated service life. An ultrasonic method that characterizes the material by tracking attenuation and velocity, as a function of frequency, over time is presented. No record of service history was assumed. A simple numerical simulation, based on the

Kelvin–Voigt model, guided by a scalar expression, matched experimental attenuation over the limited transducer frequency band and provided a viscoelastic damping coefficient, which is a measure of viscosity. Ultrasonic nondestructive evaluation was used on a sample of inert propellant at the 5th and 11th months after casting. A matched pair of 50 kHz transducers, wavelength in the material of 3.2 cm, were used in 120 tests. The preliminary experimental results indicate a decreasing attenuation and associated viscoelastic damping coefficient as time increases. Further testing will be used to ascertain an empirical relationship between attenuation, viscoelastic damping coefficient, and Young's modulus over time, and their relationship to the safe life for motors.

3:45

2pSA7. An optimized finite-difference scheme for high-loss viscoelastic materials. Mark A. Hayner and J. Robert Fricke (Dept. of Ocean Eng., Rm. 5-218, 77 Massachusetts Ave., Cambridge, MA 02139)

In an effort to attenuate helical wave energy in cylindrical shells and reduce its coupling to the surrounding fluid, various constrained layer shell designs are being investigated. The shell skins being considered have an effective loss factor η of order unity. As part of this effort, an efficient time-domain finite-difference scheme is being developed with the ultimate objective of constructing 3-D models of such shells. Optimization of relaxation time spectra for low- η materials, done by Blanch *et al.* [Technical Report, Rice University, Dept. of Geology and Geophys. (1993)], is extended to high η by matching both the real and imaginary parts of the complex modulus with experimental data. To improve scheme accuracy further, the matching is done again after discretization to minimize the effects of numerical dispersion and dissipation. Data has been matched for a material with $\eta=0.5$ over one frequency octave using a Crank–Nicholson scheme with two relaxation times. The general applicability, accuracy, and efficiency of this numerical optimization process are subjects of continuing research. Comparison between numerical simulation and analytical solutions are given for wave propagation in a constrained layer flat plate. [Work supported by ONR.]

4:00

“ROUND TABLE” DISCUSSION:

Moderators: Wayne T. Reader and Bruce Hartmann

The objective of this “Round Table” discussion period is to involve the *speakers and the audiences* of the two sessions “Characterization of viscoelastic materials I and II” in an open discussion of topics important to researchers concerned with the characterization and applications of viscoelastic materials. Suggested discussion topics include, but are certainly not limited to: measurement techniques and apparatuses, criterion for the selection of standard polymers, time/temperature shift algorithms, determination of static moduli, identification of new research areas, and clarification of issues raised in papers presented during the two sessions.

Session 2pSP

Speech Communication: Topics in Speech Recognition

Joan Bachenko, Chair

AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, New Jersey 07974

Contributed Papers

1:00

2pSP1. Use of explicit duration models in speech recognition. Cynthia Fong and Mari Ostendorf (ECS Dept., Boston Univ., 44 Cummings St., Boston, MA 02215)

Although researchers have observed that automatic speech recognition errors are often associated with improbable duration patterns, current recognition systems take little advantage of known durational cues. In addition to using simplistic models of duration, recognizers put very little weight on durational probabilities (because of the high dimensionality of the feature vectors typically used). This work explores the benefits that can be gained in speaker-independent continuous speech recognition from more accurate duration modeling and automatically estimating a duration score weight. In the context of the stochastic segment model, segment durations are modeled using gamma distributions conditioned on phonetic, lexical, and phrasal context, as well as speaking rate. Robust parameter estimation, a potential problem with the large number of different conditioning factors, is achieved using automatic clustering techniques with a maximum likelihood criterion. A by-product of the clustering procedure is a better understanding of the relative importance of the conditioning factors. Speech recognition experiments on the resource management task show that simple context-independent gamma distributions give similar results to nonparametric relative frequency models, and that error can be reduced by more than 10% by conditioning models on such factors as prepausal position, speaking rate, and lexical stress.

1:15

2pSP2. A corpus-based model of interstress timing and structure. Joan Bachenko and William A. Gale (AT&T Bell Labs., 600 Mountain Ave., Murray Hill, NJ 07974-0536)

Studies of interstress intervals tend to be more suggestive than conclusive because they rely on relatively few speech samples. The study reported here is based on observations taken from 32 000 intervals in the read speech of 106 speakers. A phone recognizer was used to label the onset times of each phone; intervals were identified as the span between stressed vowel onsets and each interval was classed according to its structure (the number of consonants and reduced vowels it contained) as well as duration. The data showed strong regularities in the dependence of time on classification. A model mixing duration, interval structure, and prior probabilities was then constructed and tested on phone lattices; the lattices were generated by the phone recognizer for speech from the resource management task. When durations were fixed but interval structure varied, prior probabilities pruned incorrect answers significantly better than chance; the mixed model's improvement was inconclusive. However, when both durations and interval structure were varied, the mixed model was clearly superior. The results indicate that interval duration and structure might be a link between speech and grammar in human and machine speech recognition.

1:30

2pSP3. Speech recognition using hidden Markov models with multiple-track polynomial regression functions as nonstationary states. M. Aksmanovic and L. Deng (Dept. of Elec. and Comput.

Eng., Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada)

The formulation of the hidden Markov model (HMM) has been successfully used in automatic speech recognition for almost two decades. In the standard formulation, the individual states in the HMM are each associated with a generally distinct but stationary stochastic process. This makes the standard HMM inadequate for representing the nonstationary property of the many speech segments intended to be described by the HMM-state statistics. A generalized HMM has been developed to overcome this inadequacy by introducing state-dependent polynomial regression functions on time that serve as the time-varying means in the HMM's Gaussian output distributions [e.g., L. Deng, *Signal Process.* **27**, 65-78 (1992)]. Recently, Aksmanovic and Deng extended the above model so that the state-dependent nonstationary process contains multiple tracks of the polynomial functions. This new parametric class of nonstationary-state HMMs has been implemented and evaluated. Experiments on fitting models to speech data, on limited-vocabulary word recognition, and on phonetic classification demonstrated advantages of the nonstationary-state HMMs over the traditional stationary-state HMMs. Details of the model implementation and of the experimental results will be described. In particular, the focus will be on comparisons between uses of single-track and multiple-track regression functions defined within the HMM states, and on comparisons among uses of varying orders of the state-dependent polynomial regression functions.

1:45

2pSP4. Duration modeling with hidden Markov models. L. F. M. ten Bosch, X. Wang, and L. C. W. Pols (Inst. for Phonetic Sci., Univ. of Amsterdam, Herengracht 338, 1016 CG Amsterdam, The Netherlands)

In hidden Markov modeling (HMM) of speech signals, the statistics of speech characteristics are represented by HMM parameters after the HMM training. This procedure is purely statistical. This study concerns the incorporation of explicit knowledge into the HMM training. Therefore one specific parameter, i.e., segment duration, was selected. In order to study the relation between duration and HMM modeling, three types of duration PDFs (DPDFs) are distinguished: (A) the DPDF defined by the segmented database used (the actual duration histogram); (B) the DPDF defined by the trained Markov model (i.e., by the transition matrix), and (C) the DPDF based on the HMM segmentation. While PDF (A) is based on data and PDF (B) is based on the trained model, PDF (C) combines both features and is based on the available set of observation sequences. First, an explicit relation is formulated between topology of the PLU, the three DPDFs, and the so-called Padé expansion. By using the *generating function* of the PDPT, it is possible to relate topological properties of PLUs on the one hand and algebraic properties of the DPDF on the other. Second, relations between those PDFs are presented by using two databases containing identical texts, but read aloud with a normal and fast speaking rate. This procedure allows a comparison between variations in the phonetic segment duration and the HMM parameters.

2pSP5. High-resolution and efficient multiple-string hypothesization using interword models. Wu Chou, Tatsuo Matsuoka, Biing-Hwang Juang, and Chin-Hui Lee (AT&T Bell Labs., 600 Mountain Ave., Murray Hill, NJ 07974)

A new accurate string hypothesization algorithm to find multiple-string hypotheses for speech recognition is proposed. The algorithm differs from the conventional N -best search algorithms in that it allows the use of the same long-term language (bigram) models and the same set of subword models, including the interword models, to perform both forward tree search and backward tree-trellis stack decoding. The proposed A^* -based backward tree-trellis stack decoding can handle interword units at the word boundaries, including the word boundaries of one-phone words. Therefore, the interword context dependency is exactly preserved in both forward and backward multiple-string hypotheses search. The search efficiency is maximized by applying the same high-resolution acoustic and language models in both search directions. When the search heuristics are used, the proposed approach provides a more accurate string model matching than that of the conventional time-synchronous beam search decoder. The proposed algorithm was tested on 24 000 connected digit strings recorded over the telephone network and collected from ten regions across the United States. Using a set of interword subword unit models, the test results showed that 62.4% string errors can be corrected if the second-best string hypothesis can be used to replace the wrong strings. The string error rate was decreased from 2.75% (657) to 1.0% (247) using the top two string hypotheses.

2:15–2:30 Break

2:30

2pSP6. Context-dependent acoustic subword modeling for connected digit recognition. Chin-Hui Lee, Wu Chou, Biing-Hwang Juang, Lawrence R. Rabiner, and Jay G. Wilpon (AT&T Bell Labs., 600 Mountain Ave., Murray Hill, NJ 07974)

Accurate and robust connected digit recognition is essential for a wide range of telecommunication services. Based on training and testing using only clean network digit data, and using the same whole-word model architecture as in the TI/NIST connected digit testing, the string error rate increased from less than 1% to more than 5%. The performance degraded even further when evaluated on data collected with different network conditions. Most of the observed errors were caused by changing channel characteristics, highly variable digit pronunciations, and inadequate modeling of cross-digit coarticulation. Results are presented for a number of context-dependent whole-word and subword modeling techniques developed to overcome some of the above problems. The most effective one is a new acoustic subword modeling approach that assumes that each digit model consists of three parts, namely, *head*, *body*, and *tail* subword units. Multiple heads and tails are also allowed, one for each of the 11 possible preceding and following digits and the background. Cross-digit coarticulation is modeled by connecting the pair of digits through the corresponding tail and head units. Testing on about 12 000 digit strings, collected from five regions, this new model architecture reduced the string error rate to under 2%.

2:45

2pSP7. On the automatic classification of pitch movements. L. F. M. ten Bosch (Inst. for Perception Res./IPO, P.B. 513, Eindhoven, The Netherlands)

The design of an algorithm that classifies pitch movements is discussed. The algorithm consists of two steps: (a) a training phase, which is based on a labeled training corpus of 249 grammatical Dutch sentences, and (b) the recognition phase. The IPO labeling system

[J.'t Hart *et al.*, *A Perceptual Study of Intonation. An Experimental-phonetic Study to Speech Melody* (Cambridge U.P., Cambridge, 1990)] is used. The setup of the algorithm is based on (A) the extraction of two time-varying features (pitch, and a feature called vowel strength), followed by (B) a search for characteristic movements over time in the resulting feature set, (C) a decision procedure similar to a linear discriminant analysis, and (D) the use of an intonation grammar. Without using (D), a classification rate of 81% is attained by using the database of 249 semispontaneous sentences, spoken by more than 40 male and female voices. Invoking the grammar [step (D)], at least 6% of the remaining 19% errors can be resolved by disambiguating phonetic candidates. The approach toward the automatic recognition of pitch movements is essentially different from the approach followed in speech segment recognition systems based on hidden Markov modeling. This difference will be discussed. Although the algorithm is developed for Dutch speech material and for the labels as defined in the IPO intonation labeling system, the method is generally applicable to other languages and to other (consistent) intonation labeling systems.

3:00

2pSP8. Artificial conversational speech signal generation method for measuring characteristics of devices operated by speech signals. Hiromi Nagabuchi, Kanako Satoh, and Nobuhiko Kitawaki (NTT Labs., Midori-cho, Musashino-shi, Tokyo, 180 Japan)

A method for generating an artificial conversational speech signal as the input test signal in measuring the characteristics of devices operated by speech signals, such as speech detectors, voice switches, and echo controllers is proposed. A state transition model among talkspurt, pause, double-talk, and mutual silence in conversational speech signals was introduced to simulate the statistical characteristics of these states in real conversational speech signals. This model assumed that the cumulative distribution of the duration time in each state is exponential. During talkspurt intervals, an artificial voice was generated using a speech synthesizer that controlled the spectrum and source characteristics independently. This artificial signal generation method can generate a signal with given characteristics by changing the values of system parameters, such as the average duration time of each state, transition probability from one state to another, and LPC (linear predictive coding) parameters that express short-term spectrum variation. The effectiveness of this artificial signal was demonstrated when used to measure the characteristics of a speech detector.

3:15

2pSP9. A speech separation algorithm that is robust to reverberation. Philip N. Denbigh and Hongyue Luo (School of Eng., Univ. of Sussex, Falmer, Brighton BN1 9QT, UK)

An algorithm has been developed that can segregate overlapping voices by exploiting harmonicity. It is primarily a monaural method that relies on tracking pitch during the voiced parts of speech, and it does this successfully in spite of the spectral modifications imposed by room reverberation. The incoming waveform is divided up into frames that undergo spectral analysis by means of the FFT. Pitch determination is based on the subharmonic summation method, but tracking constraints are imposed that improve performance. First, the continuity of pitch is exploited by limiting the search for harmonics to the regions corresponding to the harmonics of the previous pitch measurement. Second, a condition is imposed whereby the amplitude of the harmonics cannot change significantly between frames. The combination results in a "two-dimensional harmonic sieve" that segregates voiced sounds. A remaining problem is that of associating a separated voice with the correct speaker when tracking is lost, for example, after a gap in voicing or after the pitch frequencies cross. For this, a binaural cue is invoked that relies on the average difference of arrival time of the relevant harmonics of one voice between two microphones. This also proves to be robust to reverberation.

Session 2pUW

Underwater Acoustics: Seabed Acoustics

F. Michael Pestorius, Chair

Applied Research Laboratories, University of Texas, Austin, Texas 78713-8029

Chair's Introduction—12:55

Contributed Papers

1:00

2pUW1. Effects of "realistic" geoacoustic parameters on frequency-dependent bottom loss. Robert D. Stoll (Lamont-Doherty Earth Observatory of Columbia Univ., Palisades, NY 10964)

Recent field studies have shown that seismic velocity and attenuation gradients vary rapidly in the sediments immediately below the seafloor. These variations have an important effect on the frequency-dependent bottom loss that is predicted for waves entering the bottom at low grazing angles for several reasons. The more obvious reason is the variable path length traversed by the shallow diving waves that are returned to the water column; however, a second important reason appears to be the conversion of some p -wave energy to s waves resulting from the weak coupling between the two wave types that is caused by the high-velocity gradients near the bottom. Comparisons of bottom loss computed using the conventional constant gradient models and variable-gradient models based on recent field measurements show significant differences in bottom loss as a function of frequency. The additional effects of p - to s -wave conversion are also illustrated. [Work supported by ONR, Code 11250A.]

1:15

2pUW2. Seismoacoustic beamforming with a linear array of geophones on the seafloor. J. A. TenCate, T. G. Muir, J. A. Shooter, J. F. Manning (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029), Å. Kristensen, A. Caiti, and E. Michelozzi (SACLANT Undersea Res. Ctr., 19026 LaSpezia, Italy)

Experiments were carried out in the Straits of Sicily, Malta Channel where the seafloor is a relatively flat silty sediment some 150 m deep. A linear array of ten vertically gimbaled geophones with element separations of 5 m was laid out on the seafloor. Measurements were made on seismic fields created by explosive shots. Similar experiments were performed in Vestfjord, Norway, where the seafloor is mud over glacial till. The seismic waves of primary interest at these sites are the dispersive Scholte waves. Group velocity dispersion curves from several different shots were derived using the multiple-filter method and show both sites to be isotropic. From these curves the phase velocity as a function of frequency was obtained. A beamforming algorithm was implemented that accounted for the change in seismic velocity with frequency. This enabled the array to be beamsteered on the shot signals providing array responses at various frequencies. The resulting directivities and sidelobe characteristics were in good qualitative agreement with array theory. Since seismic velocities were quite low (50–130 m/s), narrow beams were formed at low frequencies; half-power widths at 18° at 7 Hz to 13° at 11 Hz were obtained with an array only 50 m in length. [Work supported by Defense Advanced Research Projects Agency, Maritime Systems Technology Office.]

1:30

2pUW3. Low-frequency acoustic transmission at a shallow water site in the Gulf of Mexico. Peter G. Cable (BBN Systems and Technol., Union Station, New London, CT 06320-6147)

During Area Characterization Test I (ACT I), conducted in September 1992 in 100-fathom water on the West Florida Shelf, acoustic transmission measurements were made covering the band 50 Hz to 1 kHz using explosive charges detonated at 90-m depth. The test site had a sand-silt-clay bottom with a gentle slope of 0.135°, and up, down, and cross-slope transmission data to ranges of 40 km were collected on collocated 25-element vertical and horizontal hydrophone line arrays. The low sea states and downward refracting acoustic conditions prevalent during ACT I ensured that transmission loss was controlled by geometric spreading and bottom interactions. Transmission loss as a function of range, frequency, and receiver depth has been determined and compared favorably with predictions for both the flat and sloping bottom cases. In addition, the multipath temporal structure and the vertical arrival structure of the received signals have been analyzed and have indicated the role of bottom and subbottom characteristics to the properties of the received signal. These analyses of the signal structure have highlighted transmission mechanisms not readily apparent in transmission loss data and have emphasized the importance of knowing subbottom structure to a detailed understanding of shallow water transmission over sand bottoms. [Work supported by the ARPA Adverse Environments Program.]

1:45

2pUW4. Complex effective depth and mode eigenvalues for a layered elastic waveguide. C. T. Tindle and N. R. Chapman (Defence Res. Establishment Pacific, FMO Victoria, BC V0S 1B0, Canada)

The complex effective depth [Z. Y. Zhang and C. T. Tindle, *J. Acoust. Soc. Am.* **93**, 205–213 (1993)] has been investigated for a shallow water waveguide with a layered elastic bottom. For multiple layers, both the complex effective depth and the reflection coefficient for the bottom have oscillatory structure as a result of resonant effects in the layers of the sediment. Under these conditions and for elastic layers, some of the normal mode eigenvalues can be difficult to find, as a simple mode counting procedure is not available. The complex effective depth leads to a formulation of an eigenvalue problem in terms of a phase integral. The phase integral is a function of complex wave number and a normal mode exists whenever it is a multiple of π . A path exists in the complex plane along which the phase integral is real and monotonic. The path passes through all eigenvalues for both trapped and leaky modes and these can be found systematically. [Work supported by DND.]

2:00

2pUW5. Influence of geoacoustic modeling on predictability of low-frequency propagation in range-dependent, shallow-water environments. R. J. Cederberg, W. L. Siegmann, M. J. Jacobson (Rensselaer Polytech. Inst., Troy, NY 12180-3590), and W. M. Carey (Advanced Res. Projects Agency, Arlington, VA 22203-1714)

In a previous study of predictability of relative intensity and hori-

zonal wave numbers by the authors, parabolic approximations were used in range-independent shallow-water waveguides. Uncertainties in sediment properties were found to be the most significant factor limiting prediction accuracy. In this paper, models of bottom sound-speed profiles that take into account sediment consolidation are developed, and their effects on propagation model predictions in range-dependent environments are examined. Environmental parameters correspond to those near the site of a recent New Jersey Shelf experiment site. Using borehole density data and Biot–Stoll theory, a functional form for porosity in a homogeneous consolidated sediment is derived. Corresponding sound-speed profiles for different sediment types are then constructed. Predictions from models consisting of consolidated sediment layers throughout the bottom are compared with results from cases for which sound-speed is comprised of piecewise linear segments. In addition, bottom sound-speed range dependence was modeled with discrete as well as continuous changes between profiles for variations of the geophysical parameters in this area and no substantial difference was observed. Comparison of experimental data with model predictions incorporating range dependence are discussed. [Work supported by ONR.]

2:15–2:30 Break

2:30

2pUW6. Simple 3-D Gaussian beam calculations for a standard and a modified wedge. Homer Bucker (NCCOSC, RTDE DIV, San Diego, CA 92152-5001)

In many practical Navy acoustic systems, the sound field is strongly affected by three-dimensional variations in the ocean bottom. A theory is presented for a simple Gaussian beam propagation model that accounts for full three-dimensional effects of the ocean bottom. The Gaussian beam formulation is especially useful because it defines the sound field in terms of the travel times and angles of eigenrays that are needed for the analysis of acoustic systems. Examples will be shown for the standard wedge calculation [Deane and Buckingham, *J. Acoust. Soc. Am.* **93**, 1319–1328 (1993)] and for a wedge modified by an island and a knoll.

2:45

2pUW7. Parabolic equation average propagation in the presence of rapid range variation in a sediment layer. Roger M. Oba (Acoust. Simulation, Code 7181, Naval Res. Lab., Stennis Space Center, MS 39529-5004)

The routine for computing the average solution to the generalized parabolic equation [R. M. Oba, *J. Acoust. Soc. Am.* **90**, 2300(A) (1991)] in the water works for propagation with sediments modeled by fluid layers. It can be used in a “marching algorithm” from one range step to another, especially if they are statistically independent. Because the algorithm relies upon analytic forms of matrices for horizontal propagation of vertical modes, a natural limit exists when (a) the sediment variation is rapid compared with wavelength and depth, (b) the sediment variation is statistically independent in each range step, and (c) the probability distribution over the velocity profile ensemble in each range step is identical. Its application to the case where the probability distribution can be parametrized by one variable is demonstrated. This

approach has advantages over averages of multiple runs of deterministic models over realizations of sediment velocities. Features of the average solution transmission loss and phase change, preserved by this algorithm, will be discussed. [Work supported by ONR/NRL-SSC and ONR Young Navy Scientist Program.]

3:00

2pUW8. Two-dimensional modeling of low-angle backscatter from geologically realistic seafloors. R. A. Stephen and S. A. Swift (Dept. of Geol. and Geophys., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The numerical scattering chamber has been used to quantitatively predict low-angle backscatter coefficients for a suite of seafloor models. A number of issues have been addressed: (a) How finely does seafloor bathymetry need to be resolved in order to predict the backscattered field? Corollary issues in this area are (i) at what resolution can one go from a deterministic to a stochastic representation and (ii) how should one interpolate between bathymetric data points? (b) How important are shear (rigidity) properties of the sediment and basement in a predictive model of seafloor backscatter? (c) How important is intrinsic attenuation (anelasticity) in a predictive model of seafloor backscatter? (d) How do the magnitude and mechanisms of backscatter vary with increasing sediment cover? (e) How effective are isolated facets on the seafloor in generating strong backscatter? (f) Can realistic volume heterogeneities within the sediment and basement generate comparable backscatter levels to realistic seafloor roughness? These issues will be discussed by comparing backscatter coefficients for various seafloor representations.

3:15

2pUW9. Long-range continuous-wave and time-domain simulations of ocean acoustic scatter from a subbottom anelastic corner. Stanley A. Chin-Bing (Naval Res. Lab., Stennis Space Center, MS 39529-5004), Joseph E. Murphy, and Gongqin Li (Univ. of New Orleans, New Orleans, LA 70148)

Recent long-range acoustic reconnaissance experiments done in the mid-Atlantic ridge region show strong reverberation from many ocean bottom features. Two types of features characteristic of this region, and possible contributors to this reverberation, are seafloor ridge corners—“inner” corners and “outer” corners—formed by two ridge faces intersecting at near-right angles. Corners formed by faces intersecting at near 90° are termed “inner” corners and those formed by faces intersecting at near 270° are termed “outer” corners. Via computer simulations, the effects of an anelastic seafloor corner on the acoustic field originating from a distant source have been investigated. Both an “inner” corner and an “outer” corner were used in the study. The effects of thin sediment layers covering the corners were also investigated. Simulations were made using continuous-wave (cw) computer models: the parabolic equation model, FEPE, for long-range propagation of the acoustic fields to and from the vicinity of the corners, and the seismoacoustic finite-element model, SAFE, to determine the scattered fields from the anelastic corners. Time-domain calculations were obtained from FFTs of the cw fields. Examples will be presented that show the effects from these two corner types. [Work supported by NRL, ONR, and the ONR Acoustic Reverberation SRP.]

Equipment Exhibit

See pages xxxi and xxxii for the list of exhibitors.

WEDNESDAY MORNING, 6 OCTOBER 1993

TERRACE ROOM, 8:45 A.M. TO 12:00 NOON

Session 3aAO

Acoustical Oceanography: Acoustical Determination of Ocean Parameters and Processes

Jeffrey A. Nystuen, Chair

NOAA-AOML, Acoustics Division, 4301 Rickenbacker Causeway, Miami, Florida 33149

Contributed Papers

8:45

3aAO1. Underwater sound produced by rainfall: Secondary splashes of aerosols. Jeffrey A. Nystuen (Ocean Acoust. Div./AOML, 4301 Rickenbacker Cswy., Miami, FL 33149) and Herman Medwin (Naval Postgraduate School, Monterey, CA 93943)

Earlier studies have identified three sources of underwater sound production from raindrops: the initial impact, a bubble trapped underwater as the impact crater closes (type I), and a bubble trapped underwater by a turbulent jet associated with the splash canopy formation (type II). For natural rainfall, type I bubbles are associated with small raindrops (0.8–1.1 mm diam), while type II bubbles are associated with large raindrops (diameter greater than 2.2 mm). Predictions for the sound generated by rainfall considering these three sound source mechanisms have been consistently low, suggesting that an additional source mechanism is present. Experimental data will be presented describing a new mechanism of bubble entrapment—namely, bubbles created during the secondary splashes of drop aerosols thrown up during the initial raindrop impact (type III). For the raindrop sizes studied (large raindrops, 3.0–4.7 mm diam), type III bubbles occurred, on average, more than once per impact. These bubbles span sizes, resonance frequencies, and acoustical energy emissions comparable to the type II bubbles. Consideration of type III bubbles improves predictions of the underwater sound generated by rainfall. These predictions are needed to support the inversion of the oceanic ambient sound field to estimated rainfall rate at sea quantitatively. [Work supported by ONR and NRL/TOWS.]

9:00

3aAO2. Resonance scattering from nonspherical bubbles in water. C. Feuillade and M. F. Werby (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Submerged bubbles can be treated as air-filled inclusions in water from which scattering occurs because of the change of acoustical impedance at the interface between the two media. The T -matrix expansion technique allows for the general description of scattering from both spherically and nonspherically shaped bubbles. This method is used to study the monopole acoustical resonances of bubbles deformed into elongated axisymmetric objects (specifically, prolate spheroids and cylinders with hemispherical endcaps). The results confirm that the reso-

nance frequency of a bubble increases when it is deformed from a spherical shape, but show that it is also independent of the direction of excitation and increases more quickly with aspect ratio for cylindrical than for prolate spheroidal bubbles. The frequency width of the resonance increases with the deformation in both cases, but Q falls more quickly for the cylindrical bubble. Increasing the deformation causes the scattering amplitude to decrease and its angular distribution to change. A remarkable result is that for highly deformed spheroidal bubbles there is relatively little change from the spherical scattering pattern obtained with an undeformed bubble. In contrast, highly deformed cylindrical bubbles show prominent lobes. [Work supported by ONR Technology Directorate (Element 602435N) and by the Naval Research Laboratory, 6.1 Program (Element 601153N). Technical management provided by NRL-SSC.]

9:15

3aAO3. Estimating vertical and horizontal wave-number temperature spectra of a buoyant thermal plume. John Oeschger (Univ. of Rhode Island, Dept. of Phys., Kingston, RI 02881) and Louis Goodman (Naval Undersea Warfare Ctr., Newport, RI 02841)

Acoustic scattering from a buoyant thermal plume described by the far-field Born approximation results in a simple relationship between the scattered pressure field and the scattering field; namely, the two are Fourier transform pairs. For media variability, such as the buoyant plume where the spatial variability is larger than the Fresnel radius, wave-front curvature must be included to describe experimental results adequately. Under conditions where Taylor's hypothesis is satisfied (the time variability is entirely due to the advection at a constant velocity of spatial variability), the effects of wave-front curvature on the scattering process can be removed. Data are collected for simultaneous multiple bistatic scattering experiments in common and antiparallel scattering directions for the unstable and turbulent plume. Results include an estimation of the vertical and horizontal wave-number temperature spectra of the buoyant plume.

9:30

3aAO4. Barotropic currents, vorticity, and tides in the northcentral Pacific in summer 1987 determined from long-range acoustic transmissions. Brian D. Dushaw (A.P.L., Univ. of Washington, 1013

Large-scale depth-integrated currents, relative vorticity, and tides were measured in the northcentral Pacific Ocean during summer 1987 using long-range reciprocal acoustic transmissions between transceivers in a triangle approximately 1000 km on a side. Tidal harmonic constants found from the acoustically determined currents agree with those found from current meters and with the tidal models of Schwiderski [E. W. Schwiderski, *Mar. Geod.* **3**, 161–255 (1980)] and Cartwright *et al.* [D. E. Cartwright *et al.*, *NASA Tech. Mem.* 104578 (1992)]. Sum travel times were used to calculate the baroclinic tide isotach displacement. A significant part of the derived internal tide is deterministic. Currents are calculated using the topographic Sverdrup balance with the Fleet Numerical Oceanography Center wind field. The measured time derivative of the areal-averaged relative vorticity is insignificant to the Sverdrup balance. Currents and vorticity calculated using the Sverdrup balance are an order of magnitude smaller than the observations. The magnitude and variability of the large-scale currents and vorticity determined from the Semtner–Chervin eddy-resolving model of ocean circulation [A. J. Semtner and R. M. Chervin, *J. Geophys. Res.* **93**, 15 502–15 522 (1988)] are similar to the direct measurements. [Work supported by NSF and ONR.]

9:45

3aAO5. Determining elastic seafloor parameters in shallow water using ambient noise. Grant B. Deane, Nicholas M. Carbone, and Michael J. Buckingham (Marine Phys. Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238)

The ambient noise field in shallow water contains information about the seafloor. Measurements of the coherence of ambient noise have been used successfully [M. J. Buckingham and S. A. S. Jones, *J. Acoust. Soc. Am.* **81**, 938–946 (1987)] to infer the critical angle of a variety of fluid basements. This technique has been extended to include the case of a shear-supporting bottom. A theoretical model of the ambient noise in an isovelocity, shallow water channel overlying an elastic basement with independent, randomly distributed surface sources will be presented. The model shows that elastic basements with shear speed greater than 650 m/s have sufficient effect on the noise field to enable the geoaoustic parameters of the bottom to be determined. The two main effects observed are an asymmetry in the vertical directionality and the arrival of energy at angles greater than the critical. [Work supported by ONR.]

10:00

3aAO6. Angular dependence of acoustic backscatter from simulated anisotropic Goff–Jordan seafloor relief. V. Premus and D. Alexandrou (Dept. of Elec. Eng., Duke Univ., Box 90291, Durham, NC 27708-0291)

The angular dependence of 12-kHz acoustic backscatter from simulated anisotropic centimeter-scale bottom relief is investigated for angles of incidence between 0° and 20° . Numerical simulations of backscattering strength as a function of elevation and azimuth are obtained by merging the Kirchhoff approximation with realizations of seafloor topography derived from the Goff–Jordan surface model. The surface model has been theoretically extrapolated to the small-scale roughness regime by connecting the spatial sampling interval with the incident acoustic wavelength and defining the characteristic wave numbers, k_x and k_y , of the 2-D surface roughness spectrum to be $O(m^{-1})$. The viability of extending the Goff–Jordan model to centimeter-scale relief is supported by the recent observation of power-law spectra within the spatial frequency range of 0.01 to 1.0 cycles/cm [D. R. Jackson and K. B. Briggs, *J. Acoust. Soc. Am.* **92**, 962–977 (1992)]. The results demonstrate that backscattering strength can be very sensitive to azimuthal variation in surface correlation properties. It is also observed that the spectral roll-off parameter has an important impact on the minimum degree of anisotropy that can be identified by the backscattered acoustic

10:15–10:30 Break

10:30

3aAO7. High-resolution backscatter from the Mid-Atlantic Ridge. Nicholas C. Makris, Jonathan M. Berkson, and Richard Menis (Naval Res. Lab., Washington, DC 20375)

High-resolution backscatter is charted to respective scattering sites on the western flank of the Mid-Atlantic Ridge (MAR) by two-way travel-time analysis and beamforming. Since the range resolution of the low-frequency FM acoustic data is on the order of the acoustic wavelength (roughly 6 m), a bistatic mapping procedure is employed that accounts for bathymetric variation and combines calculations of eigenray paths and slant range to precisely locate scattering sites at ranges up to roughly one-half a convergence zone from the source/receiver. Right–left ambiguity of the horizontal-line-array data is resolved by taking advantage of the environmental symmetry-breaking properties of transmission loss, modeled with the parabolic equation, as well as by inversion of adjacent quasimonostatic observations. Mapped reverberation and estimated scattering strength are then projected onto high-resolution bathymetry for comparison. For the same scattering site and observation geometry, it is found that higher-resolution signals often yield higher and more localized scattering strength maxima. This suggests that discrete and spatially intermittent scatterers are important to reverberation in the MAR.

10:45

3aAO8. Model eigenfunction perturbations and group speed tomography. B. Edward McDonald (Naval Res. Lab., Washington, DC 20375) and Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

Certain problems in acoustic thermometry of ocean climate (ATOC) require evaluating the response of modal group speeds to perturbations in the ocean sound-speed profile. Two different approaches have been taken recently by researchers, resulting in two perturbation expressions whose equivalence is not immediately evident. The first approach, used by Lynch *et al.* [*J. Acoust. Soc. Am.* **89**, 648 (1991)], results in an integral equation involving the frequency derivative of the eigenfunctions. The second approach, under current investigation, yields an integral equation involving the response of the eigenfunctions to sound-speed perturbations. The two expressions are shown to be equivalent by means of their projections onto the set of modal eigenfunctions. These results are discussed in relation to group travel-time tomography.

11:00

3aAO9. Effects of perturbations on long-range propagation. William A. Kuperman,^{a)} Michael D. Collins, B. Edward McDonald (Naval Res. Lab., Washington, DC 20375), and Kevin D. Heaney^{a)} (Planning Systems, Inc., McLean, VA 22102)

Results of numerical modeling of long-range (greater than 500-km) low-frequency acoustic propagation in the ocean have been presented previously. These results were in qualitative agreement with measurements for the Heard Island Feasibility Test and the SLICE89 experiments. To quantify the differences between experimental and parabolic equation (PE) results, time-dependent modal decomposition was applied to the received signals. This modal tomography gives us an indication of the depth-dependent differences between the actual and assumed sound-speed profiles. This work is extended to look at the effects of small perturbations in the sound-speed field (i.e., internal waves) on the arrival pattern. Internal waves were modeled using a statistical realization of the Garrett–Munk spectrum. Results of the effects of internal waves on specific deep ocean mode functions and model group velocities are presented. ^{a)}Present address: Marine Physical Labs., Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093.

3aAO10. Global acoustic mapping of ocean temperatures (GAMOT). John L. Speisberger (Dept. of Meteorol. and the Appl. Res. Lab., 512 Walker Bldg., Penn State Univ., University Park, PA 16802), Dan Frye (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Harley Hurlburt, Joe McCaffrey (Naval Res. Lab., Stennis Space Center, MS 39529), Mark Johnson (Univ. of Alaska, Fairbanks, AK 99701), and James O'Brien (Florida State Univ., Tallahassee, FL 32306)

Successful feasibility experiments for mapping global ocean warming and climate variability with sound have been conducted since 1983 [J. L. Speisberger *et al.*, *J. Acoust. Soc. Am.* **92**, 384–396 (1992)]. The GAMOT group is developing two new high-technology instruments for making these same measurements in near-real time at one-eighth the cost of our previous measurements taken from sources and receivers connected to shore with cables. One instrument is a surface suspended acoustic receiver (SSAR), which dangles a hydrophone array beneath a freely drifting surface unit. The other instrument is a subsurface autonomous mooring equipped with an acoustic source and a new telemetry scheme for removing travel time changes due to mooring wander in real time. Climatic temperature variability from past and present experiments is studied with state-of-the-art ocean models. A program update will be presented. [Work is supported by the Advanced Research Projects Agency.]

11:30

3aAO11. Error analysis of ocean acoustic tomography due to internal waves. J. Gozani and L. Boden (Cooperative Inst. for Res. in Environmental Sci./NOAA, Campus Box 449, Boulder, CO 80309-0449)

A realistic error analysis of the temperature variability of the oceans as probed by ocean acoustic tomography is presented. Travel-time errors resulting from fluctuating internal and surface waves are important

in the analysis of the averaged time delays, which may indicate global warming trends due to the greenhouse effect. The 3-D propagation models through random media are extremely complicated. Propagation of model equations is governed by a set of matrix differential equations, whose elements are obtained by multidimensional oscillatory integration. Analytical and numerical studies have been conducted that show the feasibility of computing the matrix elements 1000 times faster than was previously done. The matrix equation is solved by well-known ordinary differential equation methods. The effects of frequency, trajectory, sound-speed profile, bathymetry, and the variances of internal and surface waves will be discussed.

11:45

3aAO12. Sound velocity in the deep sea: Examination of measurement data obtained from a deep submergence vehicle. Toshio Tsuchiya, Yasutaka Amitani, Hiroshi Ochi (Deep Sea Technol. Dept., Japan Marine Sci. and Technol. Ctr., 2-15 Natsushima-cho, Yokosuka, 237 Japan), and Toshiaki Kikuchi (Natl. Defense Acad., 1-10 Hashirimizu, Yokosuka, 239 Japan)

An accurate equation of sound velocity in seawater is a very important factor for ocean acoustic tomography according to acoustic travel time across the ocean. A sound velocity equation has been examined in a laboratory under various values of temperature, salinity, and pressure by Wilson, DelGrosso, Chen, and Millero. However, as direct measurement in the deep-sea area is very difficult, these data have been very limited. Therefore, recently the sound velocity was measured directly more than 50 times from the surface to a depth of over 6000 m by a sing-around sound velocimeter installed in the manned deep-sea research submergence vehicle "SHINKAI 6500" of the Japan Marine Science and Technology Center. Data were obtained with temperature, salinity, and pressure values in the western Pacific Ocean area. Then, the measured data were compared with the calculated values of the equation of Wilson, DelGrosso, Chen, and Millero. The comparison result indicated a trend: Sound velocity in the deep sea is greater than values of various equations according to the pressure effect.

WEDNESDAY MORNING, 6 OCTOBER 1993

SPRUCE ROOM, 9:00 TO 11:15 A.M.

Session 3aEA

Engineering Acoustics: Acoustic Measurements and Instrumentation

Edward F. Rynne, Chair

Naval Command, Control and Ocean Surveillance Center, RDTE Division, Code 711, 496640 Gate Road, Room 19, San Diego, California 92152-6242

Contributed Papers

9:00

3aEA1. Rayleigh's horn equation. John T. Post and Elmer L. Hixson (Dept. of Elec. and Comput. Eng., Univ. of Texas, Austin, TX 78712)

Rayleigh's work on tapered waveguides is applied to acoustic horns. The acoustic velocity potential is expanded in a perturbation-type solution and the first perturbation is shown to produce the Webster horn equation on-axis, but predicts progressive waves to have curved, or bulging, phase fronts. Applicability of Rayleigh's first perturbation is shown to be identical to the estimate given by Pierce for applicability of the Webster horn equation and both of these reveal that the Webster equation *cannot* predict a "cutoff" frequency. When operating at the "cutoff" frequency predicted by Webster, any practical horn is too short to load an acoustic source, so the performance is virtually the same as letting the source radiate into free-space; but the horn does *not* attenuate the progressive wave below the "cutoff" frequency in the same way that

high-order modes are attenuated below their cutoff frequency in an acoustic waveguide. Rayleigh's second perturbation is presented as "Rayleigh's horn equation." The acoustic impedance at the throat of the horn and sound pressure interior to the horn as predicted from Rayleigh's theory are compared with laboratory measurements.

9:15

3aEA2. High-fidelity displacement sensors for acoustic emission source characterization. C. M. Fortunko, M. A. Hamstad, and D. W. Fitting (Natl. Inst. of Standards and Technol., 325 Broadway, Boulder, CO 80303)

Acoustic emission (AE) source characterization requires the use of broadband, typically 20 kHz–2 MHz, "point-contact" sensors. Generally, AE sensors based on the NIST "conical" transducer approach can meet this requirement. However, because naturally occurring AE events

are considerably less energetic than those produced by artificial AE sources, optimization of the sensor signal-to-noise performance is of critical importance. Using a computer model, a variety of sensor-design approaches have been studied, with particular emphasis on transducer/preamplifier/specimen compatibility issues. In this paper, computer-model predictions are compared with the results of experiments. The use of different piezoelectric transducer materials—PZT-5A, lead metaniobate, LiNbO_3 , X-cut quartz, and PVDF—is considered. It is shown that ceramic materials exhibiting the highest dielectric constants, ϵ_{33}^s , are best suited for most high-performance AE sensor applications. In addition, the impact of preamplifier input capacitance, including the Miller effect, and specimen mechanical impedance on the performance of a “point-contact” sensor is discussed. In particular, using the computer model, the limitations of sensors that use piezoelectric ceramics and single-crystal materials are examined. It is shown that in certain situations, such as very thin metal plates and materials that exhibit very low characteristic impedances, sensors that use piezoelectric ceramic and single-crystal transducers may not be the best choice. Therefore, to address such cases, alternative sensor approaches are needed.

9:30

3aEA3. Modulation-frequency dependence of the nonlinear sensitivity of a Brüel & Kjær 8103 hydrophone in parametric demodulation of amplitude-modulated ultrasound. Shengke Zeng and Richard B. Beard (Biomed. Eng. and Sci. Inst., Drexel Univ., Philadelphia, PA 19104)

The interference demodulation by the nonlinearity of hydrophones cannot be ignored in parametric demodulation of amplitude-modulated (AM) ultrasound in the near-field media, where the intensity of the primary incident waves on hydrophones is very high. The interference demodulation, which is caused by the second-order sensitivity of a hydrophone, $N(f)$, is proportional to the square of the incident pressure p^2 [Moffett and Blue, Naval Underwater System Center, Tech. Memo. 801150 (1980); Humphrey and Hsu, Proceeding of Specialists Conference on Underwater Acoustic Applications (1980)]. It is found in the measurement of parametric demodulation of AM ultrasound by a Brüel & Kjær 8103 hydrophone that the $N(f)$ is not only the function of the carrier frequency f , but also the function of the modulation frequency F . At a fixed carrier frequency $f_0 = 2.3$ MHz, the interference demodulation can be observed at the modulation frequencies below 6 kHz, and becomes stronger toward low modulation frequencies at a distance of 150 mm between the transducer and the hydrophone. The interference demodulation by the nonlinearity of the hydrophone can be distinguished from parametric demodulation by observing the directivity pattern difference between them. $N(f, F)$ is observed in the above parametric demodulation to be proportional to $F^{1.32}$ [S. Zheng, Ph.D. dissertation, Drexel University (1993)]. [Work partially supported by Electro-Stim Corp.]

9:45

3aEA4. Artificial neural nets for acoustic nondestructive evaluation. Peeter Akerberg, Ben H. Jansen (Dept. of Elec. Eng., Univ. of Houston, Houston, TX 77204-4793), Shail R. Pandya, Zhijiang Wang, and Robert D. Finch (Univ. of Houston, Houston, TX 77204-4792)

Artificial neural networks (ANNs) are being used for the detection of cracks in metal and concrete beams and structures. Vibration signals are obtained from these structure by impacting with a small hammer and recording the activity of an accelerometer attached to the structure. These vibration signals are digitized, and the samples are input to an ANN. Detection of cracks proceeds in two different ways. For both methods, the networks are trained on data obtained from intact structures, but in the case of the first method, training is halted once the ANN can accurately predict future samples of the vibration signal from present and past observations. These ANNs are then fed with data obtained from structures with varying degrees of defects, and the prediction error is noted. In the second method, training continues after a small defect has been made to the structure, and the weights of the nets

are compared. With both methods, we can detect cracks as small as 0.1 in. Examples of the results obtained for metal and concrete beams will be shown. [Work supported by NSF Grant no. MSS-9024224.]

10:00

3aEA5. Ultrasonic detection of icing onset and accretion thickness on aircraft. David K. Hsu, Frank J. Margetan, Samuel J. Wormley (Ctr. for NDE, Iowa State Univ., Ames, IA 50011), and Jeffrey A. Simpson (J-Tec Associates, Inc., Cedar Rapids, IA 52401)

A major environmental hazard in aviation is ice buildup on aircraft. The development of a detector for the onset of icing and for accretion thickness measurement using ultrasonic waves is described. The detection is based on reflectivity change at a metal-ice interface and based on echo-ranging techniques using pulses of longitudinal and shear waves. The detector is sensitive to the onset of icing; ice layers as thin as 0.002 in. can be detected and accurate thickness can be measured from 0.015 in. onward. By using the ratio of an interface echo and an echo from an internal reference reflector, the quantitative measurement results are immune to changes in transducer coupling conditions. Using a combination of longitudinal and shear waves, the detector can also differentiate between ice and water. Laboratory tests were made for anticipated complications, such as rough or tilted surface of ice buildup and accretion of glaze versus rime ice. [Work supported by an Army SBIR grant to J-Tec Associates, Inc.]

10:15

3aEA6. Application of nonlinear dynamics for noise reduction in measured signatures containing broadband information. Ronald Hughes (Radix Systems, Inc., 6 Taft Ct., Rockville, MD 20850)

Nonlinear dynamics has been of great academic interest in recent years. Its concepts have been employed as a foundation for modeling nonlinear processes in diverse fields extending from economic trends to medical dysfunctions. Although the developments of nonlinear dynamics concepts in acoustics are sparse, their application in acoustics has advantages over linear techniques in extending broadband signatures in high-noise environments. Linear signal analysis approaches are significantly limited when the noise environment is nonstationary or when the signal duration is short. Furthermore, the application of filters alters the information content of the original broadband signals. Nonlinear dynamics methods do not suffer these limitations. Two such methods will be described for effecting noise reduction where the signal-to-noise ratio is zero or negative and where there is no *a priori* knowledge of either the signal or the noise. Signals used for demonstration include sinusoids and chaotic sequences; additive noise includes uniform and Gaussian random noise and noise that produces a power spectrum equivalent to that of the “clean” sequences. Noise reduction of 15–18 dB using nonlinear dynamics will be shown, and fidelity reproduction of the original signal will be demonstrated.

10:30

3aEA7. On the nonlinearity of exhaust system acoustics in firing spark-ignition engines. A. Selamet (Dept. of Mech. Eng. and Appl. Mech., Univ. of Michigan, Ann Arbor, MI 48109-2125)

An extensive amount of literature is available on the performance of automotive silencers subject to linear acoustic disturbances and non-mean or steady-mean flow conditions. The flow through the automotive exhaust system exhibits, however, a number of nonlinear phenomena including high sound pressure levels reaching 180 dB and varying mean flow, as well as spatially and temporally changing temperatures. These nonlinearities are difficult to treat with the linearized acoustic theory approach in the frequency domain. The present study provides experimental results as well as numerical predictions for a production vehicle full exhaust system (Ford 1.9L Escort engine) and investigates the nonlinearities. The study implements a time-domain finite-difference approach to predict the acoustic performance based on the work of Chapman *et al.* [Winter Annual Meeting of ASME (1982)], which solves the one-dimensional, variable cross-sectional area, nonlinear balance equa-

tions of mass, momentum, and internal energy coupled with the equation of state for compressible flows. Nonlinearities are discussed in view of the experimental data and by comparing the terms of the momentum balance from computational results.

10:45

3aEA8. The effect of vehicle exhaust system components on performance and noise in firing spark-ignition engines—An experimental study. A. Selamet, M. S. Croy (Dept. of Mech. Eng. and Appl. Mech., Univ. of Michigan, Ann Arbor, MI 48109-2125), R. A. Kach, and J. M. Novak (Ford Motor Co., Dearborn, MI 48121)

In vehicle exhaust systems, the sound attenuation and the reduction of flow losses are often competing demands. The present experimental study considers a full vehicle exhaust system and investigates both the sound attenuation and the flow performance of production configurations including the catalyst, the resonator, and the muffler. Dynamometer experiments have been conducted with a firing Ford 1.9-Liter I4 Escort engine with speeds ranging from 1000 to 5500 rpm. Measurements including the flow rates, the temperatures, and the absolute dynamic pressures of the hot exhaust gases at key locations (upstream and downstream of every component) with fast-response, water-cooled piezoresistive transducers facilitate the calculation of acoustic performance of each component, as well as the determination of flow losses caused by these elements and their influence on the engine performance. The present study describes the experimental aspects of an extensive effort toward employing nonlinear fluid dynamic models in the time

domain for the prediction of the acoustic and power performance of firing internal combustion engines with full production exhaust system.

11:00

3aEA9. Determination of the performance of sound intensity probes in standing wave fields. Erling Frederiksen and Jørgen I. Christensen (Dept. of Microphone Devel., Brüel & Kjær, 2850 Nærum, Denmark)

IEC and ANSI standards that specify the performance of instruments for the measurement of sound intensity are expected to be released in the near future. The drafted standards prescribe the minimum performance of sound intensity probes in standing wave fields. Generally, the measurement of sound intensity has the greatest interest in connection with reactive sound fields. Unfortunately, the more reactive the sound field, the higher are the requirements to the instrument for the same measurement accuracy. As well-defined reactive fields can be produced inside tubes, standing wave tubes have been selected for testing of intensity probes. The standing wave performance of intensity probes, equipped with pressure-sensing microphones, has been calculated and measured. Results obtained by the two methods will be shown. A description will be given of the calculation model and of the specially designed standing wave tube that has a frequency-independent standing wave ratio of 24 dB (± 0.5 dB) between 40 and 500 Hz. This performance has been obtained by terminating the test tube with many long and narrow plastic tubes.

WEDNESDAY MORNING, 6 OCTOBER 1993

GOLD ROOM, 8:25 TO 11:55 A.M.

Session 3aMU

Musical Acoustics: Mode Studies in Musical Instruments

Uwe J. Hansen, Chair

Department of Physics, Indiana State University, Terra Haute, Indiana 47809

Chair's Introduction—8:25

Invited Papers

8:30

3aMU1. The physics of normal modes. Gabriel Weinreich (Randall Lab. of Phys., Univ. of Michigan, Ann Arbor, MI 48109-1120)

This is primarily a tutorial paper to organize and clarify various concepts whose understanding is required for work with linear systems. Among the questions to be addressed are: What is the difference between a resonance and a normal mode? What about an antiresonance? How do normal modes relate to free motion? To forced motion? Are there "strong" and "weak" modes? Are resonance denominators quadratic in frequency, or linear? What does it mean for modes to be "coupled"? In a measured frequency characteristic, when are normal modes represented by peaks, by dips, by both? What information is carried by the phase? In searching for normal modes, what is the difference between pulse excitation, white noise excitation, and sinusoidal excitation? Are there other options of interest? Do dissipative systems have normal modes? [Work supported by NSF.]

9:10

3aMU2. The role of finite-element analysis in mode studies of musical instruments. Oliver E. Rodgers (Mech. Eng. Dept., Univ. of Delaware, Newark, DE 19711)

Vibration analysis using finite-element methods of the mechanical systems of musical instruments have been and will continue to be an important and powerful tool for research and development workers studying musical instruments. If properly planned, executed, and interpreted, a finite-element calculation can describe all of the possible vibration modes, including some that might be missed in experimental studies because of complications in exciting and recording systems. It can be very useful for making studies of the effect of single design or tuning variables on mode-frequency placement. The method is especially valuable when the instrument under investigation is made of wood, an extremely variable and unpredictable material. (Knowledge of how to adjust the frequencies of important modes is very important when assembling and fine-tuning most stringed instruments.) Examples of successful use and opportunities for further use will be described.

3aMU3. Mechanical vibrations and radiation fields of guitars. Matthew Brooke (Paul S. Veneklasen & Associates, 1711 16th St., Santa Monica, CA 90404) and Bernard Richardson (Univ. of Wales College of Cardiff, P. O. Box 913, Cardiff CF2 3YB, UK)

Some of the research undertaken by the Cardiff group to try to establish relationships between the classical guitar's construction and its acoustical output are reviewed. Real instruments have been studied using techniques such as holographic interferometry, and the structural vibrations and associated radiation fields have been modeled using finite-element analysis and boundary-element methods. Primitive psychoacoustical tests have established that radiated frequency components of plucked notes are important in recognizing individual instruments. Large amounts of data, collected over the years, have clearly demonstrated that there are no simple relationships between the modal properties of instruments and their estimated "quality." Similar conclusions have, of course, been drawn by other workers in this field. Recently, the authors have concentrated more on determining the precise role played by individual modes in coupling the strings to the body and in radiating energy to the surrounding air; it is felt that this sort of detail is more indicative of the instrument's final acoustical action. These investigations emphasize the importance of the precise shapes of modes, particularly in the vicinity of the bridge. A discussion of the ways in which the luthier can fine-tune mode shapes and hence maintain control during instrument manufacture is included.

10:10

3aMU4. Modal analysis of noncircular cylinders and bells. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115) and Uwe J. Hansen (Indiana State Univ., Terre Haute, IN 47809)

In oval cylinders, the familiar vibrational modes of a circular cylinder are split into doublets, one new mode having a node in the region of greatest curvature and the other having an antinode there. Since increasing the curvature increases the effective stiffness and thus raises the speed of bending waves, the bending wavelengths are generally longer (and the nodes further apart) in the region of greater curvature. This phenomenon is the basis of the "two-tone" phenomenon in ancient Chinese bells. The mode splitting in a variety of noncircular cylinders and bells, observed by means of holographic interferometry and experimental modal testing with impact excitation, is described. By using two mirrors, it is possible to view simultaneously the front, end, and top of a vibrating cylinder in holographic interferograms.

10:40

3aMU5. Extension of modal analysis techniques: Wind instruments and radiated sound fields. Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809) and Ingolf Bork (Physikalisch-Technische Bundesanstalt, 3300 Braunschweig, Germany)

The study of normal modes of a vibrating structure using modal analysis involves the phase relationship between a large number of coherent signal pairs obtained over a predetermined geometrical grid covering the structure. From a set of transfer functions, resonance frequencies are obtained and an animated picture of the motion at the resonance frequency is displayed on a computer screen. Extending this technique to standing and traveling waves in air involves using an excitation source as reference and forming transfer functions between that reference source and sound pressure level measurements at predetermined locations in the sound field. The technique will be illustrated with animations of air modes in a flute and with animations of sound fields radiated by a piano at several frequencies [I. Bork, *Acustica* **75**, 154–167 (1991)].

Contributed Papers

11:10

3aMU6. Brass instrument bell vibrations and coupling to air modes. Peter L. Hoekje, Colby A. Payne, and David N. Kjar (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50614-0150)

The vibrations of brass instrument bodies are important to musicians. Three proposed mechanisms by which players might detect these vibrations are radiation, mechanical coupling to hands and mouth, and alteration of the acoustic response of the instrument bore as felt by the lips. The first mechanism has been demonstrated by other researchers [e.g., B. Lawson and W. Lawson, *J. Acoust. Soc. Am.* **77**, 1913–1916 (1985)]. The mechanical coupling is strong, but this mechanism has minimal musical significance. For the third mechanism, the coupling between the instrument shell vibrational modes and the cross-sectional modes of the air in the bore is dependent on their relative symmetries. In the narrow main bore only plane waves propagate; in theory these should couple only to cylindrically symmetric shell modes. Trombone bell vibrational mode patterns, identified by holographic interferometry, exhibit symmetries similar to those in church bells, with nodal circles and nodal meridians. However, they may be indexed by the single number m of the nodal meridian planes. For $m > 0$, the frequencies f_m are approximately described by a simplified Chladni's law of the form $f_m = C(m)^p$. For a typical trombone bell, $C = 209$ Hz and $p = 1.1$, while $f_0 = 277$ Hz.

11:25

3aMU7. Tracking the dynamics of musical instruments based on a high-resolution time-frequency representation. William J. Pielemeier and Gregory H. Wakefield (EECS Dept., Univ. of Michigan, 1301 Beal Ave., Ann Arbor, MI 48109)

Attacks and other phenomena that involve rapid amplitude and frequency variations over time present a difficult problem in the study of the dynamics of musical instrument modes. Fourier-series-based methods provide adequate time resolution under the conditions of signal periodicity but performance deteriorates rapidly with gross inharmonicity or large amplitude and frequency transients. Short-time Fourier transform (STFT) methods are most robust to these variations but provide limited resolution in time and frequency. In a previous paper, an alternative time-frequency representation that achieves better resolution than STFT methods, yet remains robust to conditions degrading Fourier series analysis, was presented [Pielemeier *et al.*, *J. Acoust. Soc. Am.* **92**, 2430(A) (1992)]. The present paper develops estimators for modal parameters of musical instruments based on this alternative time-frequency representation. The performance of these estimators is analyzed and compared with that of Fourier series and STFT methods for several musical cases, including the attack transients of trumpet, vibraphone, and flute notes. In general, it is found that the proposed estimators provide a more accurate portrait of the spectral dynamics of the

musical signal than is possible using estimators based on the other two representations.

11:40

3aMU8. The perfect clarinet reed? Vibrational modes of realistic clarinet reeds. Donald Casadonte (Dept. of Music, Ohio State Univ., Columbus, OH 43210)

Using results obtained from prior investigations into the biology, chemistry, and physics of the clarinet reed [D. Casadonte, *J. Acoust. Soc. Am.* **90**, 2351(A) (1991)], a realistic computer simulation using the ANSYS finite-element program has been developed. Using this simulation, modal frequencies and shapes were isolated for those modes

below 20 000 Hz, as well as stress maps of the reed structure, etc. The simulation is flexible enough to allow for the inclusion of any type of ligature, embouchure, air column pressure spectrum, mouthpiece, and variations in reed material properties and biology during any stage of the life cycle of the reed. This allows for most questions of practical importance in reed science to be examined (including those of historical reed shapes). Specifically, the problem of reed resonance [S. C. Thompson, *J. Acoust. Soc. Am.* **66**, 1299-1307 (1979)] is investigated. It is shown that a good reed is one "tuned" to maximize statistically the power output from the overtones of the air column. This tuning is gradually lost as the reed ages. Finally, the reverse procedure of designing a reed shape and material that will maximize the resonance for a typical clarinet air column is studied.

WEDNESDAY MORNING, 6 OCTOBER 1993

DENVER ROOM, 8:15 TO 11:45 A.M.

Session 3aNS

Noise: Characterization of Environmental Noise Impact

Henning E. von Gierke, Chair
1325 Meadow Lane, Yellow Springs, Ohio 45387

Chair's Introduction—8:15

Invited Papers

8:20

3aNS1. Airport noise: The problem and some solutions and frustrations. Paul E. Tauer (Mayor, City of Aurora, Colorado, 1470 S. Havana, Ste. 808, Aurora, CO 80012)

Citizens of nearby communities, usually with little or no direct control over airport operations, bear the brunt of airport noise. Minimal concern for the impact of this noise has been shown by the airlines, the airports, and the F.A.A. Political pressure and legal action have both been used in attempts to force the parties responsible to address the problem. The author has worked for several years on these issues in his capacity as former Council Member and current Mayor of a city of 235 000 residents that borders both the soon-to-close Stapleton Airport and the new Denver International Airport, and as President of the National Organization to Insure a Sound-controlled Environment (NOISE). A significant project to reduce the impact of airport noise was the Stapleton Noise Insulation Program, which involved the expenditure of \$20,000,000 to sound-insulate homes in Aurora, CO. Recent legislation regarding Stage III aircraft will reduce noise at the source, but does not adequately address noise impact. Thus, political conflict, prolonged litigation, and airport-capacity limitations will continue. There is a solution: communication and cooperation.

8:45

3aNS2. Community annoyance by aircraft noise. James M. Fields (10407 Royal Rd., Silver Spring, MD 20903)

Rigorously designed social surveys, not public complaint actions, provide the most direct available evidence about the impact of environmental noise on residents. The balance of the available social survey evidence indicates that while personal attitudes affect noise annoyance, demographic characteristics and ambient noise levels do not affect annoyance with audible sounds. A continuously graded annoyance reaction to noise does not provide a strong scientific basis for choosing "highly annoyed" rather than any other degree of annoyance as an acceptability criterion. A rigorous synthesis of annoyance survey results must screen out erroneous data, use objective methods for selecting data sets, systematically adjust for important differences in nominal noise measurement conditions, weight data points by their precision, have a firm empirical basis for relating diverse annoyance measures, and satisfactorily evaluate the likely precision of the resulting synthesis curve. A theoretically sound justification for the form of a dichotomous dose/response curve must include a theory about the distribution of measured reactions to noise. Existing dose/response curves continue to make useful contributions to noise policy. Additional work will be needed before a synthesis of community dose/response relationships can be developed that meets the previously mentioned criteria.

9:10

3aNS3. Current status of sleep disturbance research and development of a criterion for aircraft noise exposure. Lawrence S. Finegold (USAF Armstrong Lab., AL/OEBN, 2610 Seventh St., Wright-Patterson AFB, OH 45433)

There currently exists no generally accepted criterion for an acceptable level of nighttime sleep disturbance from aircraft noise. Indeed, there is little agreement concerning the appropriate scientific definition of sleep disturbance, the appropriate noise exposure metric for this environmental effect, or the circumstances under which such predictions need to be included in

environmental impact analyses. However, with the large body of scientific evidence on sleep disturbance due to environmental noise that has accumulated over the past 20 yr, it is now possible to promote an interim sleep disturbance curve for use in the federal environmental impact analysis process. This presentation focuses on the results of recent research and the implications for environmental policy decisions. Both laboratory and field research, including research in the United States and Europe, are addressed. Development of the sleep disturbance curve incorporated into the 1992 report of the Federal Interagency Committee on Noise (FICON) is also reviewed.

9:35

3aNS4. Assessment of speech interference from aircraft flyover noise in schools and other communication critical environments. Karl Pearsons (BBN Systems and Technologies, 21120 Vanowen St., Canoga Park, CA 91303)

Although most assessments of the effects of community noise impacts on people concentrate on annoyance, it is important in some speech-critical settings to assess the degree to which noise interferes with communication as well. Speech intelligibility metrics are helpful in such situations, even though they require more detailed information about noise exposure than is captured by cumulative metrics such as the day-night average sound level (DNL). Alternate metrics for assessing speech intelligibility and their uses and limitations will be described. Comparisons will be made between such metrics and those commonly used to assess environmental noise effects on a long-term average basis.

10:00

3aNS5. The present aircraft noise evaluation process. Louise Maillett (Federal Aviation Admin. Office of Environment and Energy, 800 Independence Ave., S.W., Washington, DC 20591)

The legal background and policy decisions underlying present U.S. Government noise impact assessment procedures are outlined. Collaboration and review processes to satisfy the mandates of the various agencies are discussed, with emphasis on the FAA's mission and plans regarding aircraft noise abatement.

10:25

3aNS6. New language and processes for examining environmental noise. Nicholas P. Miller (Harris, Miller, Miller, & Hanson, Inc., 429 Marrett Rd., Lexington, MA 02173)

Guidelines and standards have been developed that identify the relationship between noise levels and land uses. These relationships were intended to define what land uses were "compatible" with specific levels of noise, but, in doing so, took into consideration not only public health and welfare, but also such concerns as the rights of property owners, available technology, and feasibility. The inclusion of these basically economic concerns meant that the noise and land-use compatibility guidelines would not necessarily protect people from all the adverse effects of noise or, perhaps more important, even identify when and where adverse effects might occur. This distinction between land-use compatibility and adverse effects on people has generally been lost or overlooked. It is time to make this distinction clear. Case histories where pursuing land-use compatibility did not protect people from the adverse effects of noise or identify these adverse effects are discussed. A language and approach are proposed for clearly differentiating between the concepts of land-use compatibility and identification of disruptive effects.

10:50

3aNS7. Suggestions for solutions. Kenneth McK. Eldred (Ken Eldred Eng., Meadow Cove, East Boothbay, ME 04544-0501)

Solutions to some of the problems that inhibit effective functioning of the environmental impact evaluation process are suggested, including approaches for the use of metrics to improve communication with the affected public so that quantitative descriptions of both single-event and cumulative noise are more nearly related to the perceptions; development of guidelines for the definition of project study areas to include all who are anticipated to experience significant changes, quantification of individual changes in noise, as well as the net populations exposed to various levels, that address concerns within the entire study area; and research to improve our understanding of noise annoyance and its alleviation and to extend our noise modeling capability to areas further from airports than presently possible.

Contributed Papers

11:15

3aNS8. The new City of Houston sound regulation ordinance. C. Moritz (Collaboration in Sci. and Technol., Inc., 15835 Park Ten Pl., Ste. 105, Houston, TX 77084-5131) and H. Huey (Councilwoman, District A, City of Houston, P.O. Box 1562, Houston, TX 77251)

In January of 1993, the City of Houston adopted a new sound regulation ordinance. Previous ordinances did not specify allowable sound levels, only that it was unlawful to make loud or excessive noises. The city health department had used values of 75 dBA (daytime) and 65 dBA (nighttime) to determine if noises were excessive. The new ordinance established allowable A-weighted sound levels of 65 dBA (daytime) and 58 dBA (nighttime) on residential property and 68 dBA (day or nighttime) on nonresidential property. Enforcement under this

ordinance is now shared between the city health and police departments. Because there is no zoning in the City of Houston, this ordinance poses some unique challenges; manufacturing facilities, bars, and nightclubs are often located in residential areas. The motivation for this new ordinance and recent experiences with its enforcement will be discussed.

11:30

3aNS9. Design of a large-scale, in-home study of noise-induced sleep disturbance. Sanford Fidell, Karl Pearsons, Richard Howe (BBN Systems and Technologies, 21120 Vanowen St., Canoga Park, CA 91303), and Lawrence Finegold (USAF, Wright-Patterson AFB, OH 45433-6573)

Pearsons, Barber, and Tabachnick (1990) have documented large

differences in the findings of laboratory and home studies of the ability of noise to disturb sleep. More recently, preliminary analyses of data collected in a large-scale, in-home study of sleep disturbance [Ollerhead *et al.* (1992)] suggest a lower probability of awakening than indicated by a dosage-response relationship recommended by Federal Interagency Committee on Noise [FICON (1992)]. The U.S. Air Force is

currently conducting an in-home study of sleep disturbance, intended in part to clarify several issues not fully resolved in prior work. These include the definition of sleep disturbance and the temporal linkage between noise exposure and sleep disturbance. The rationale, design, and progress of the Air Force study are described.

WEDNESDAY MORNING, 6 OCTOBER 1993

SILVER ROOM, 8:25 A.M. TO 12:15 P.M.

Session 3aPA

Physical Acoustics and Education in Acoustics: Frontiers of Physical Acoustics

Seth J. Putterman, Chair

Physics Department, University of California, Los Angeles, California 90024

Chair's Introduction—8:25

Invited Papers

8:30

3aPA1. Thermoacoustic engines. Gregory W. Swift (Condensed Matter and Thermal Physics Group, Mail Stop K764, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Thermoacoustic engines are energy-conversion devices that achieve simplicity and reliability by use of acoustics. Their efficiency can be a substantial fraction of the Carnot efficiency. In thermoacoustic prime movers, heat flow from a high-temperature source to a low-temperature sink generates acoustic power. In thermoacoustic refrigerators, acoustic power is used to pump heat from a low-temperature source to a high-temperature sink. Applications of thermoacoustics under development at several institutions will be shown, possibility including refrigeration, sonar, cryogen liquefaction, and electric-power generation. Fundamental principles of thermoacoustics will be reviewed, and assumptions underlying our present (linear) understanding will be discussed. Limitations of our present understanding will be outlined, and their implications for design of practical devices will be discussed.

9:00

3aPA2. Helioseismology: Listening to the inside of the sun. Jurl Toomre (JILA, Univ. of Colorado, Boulder, CO 80309)

The sun is oscillating in an intricate though gentle manner. The patterns of motion observed in the solar atmosphere are complicated, for they result from interference between about 10^7 resonant modes of oscillation of the interior. The sun possesses both acoustic p modes, for which pressure is the restoring force, and gravity g modes, for which buoyancy is the dominant force. Each resonant mode resides in a cavity beneath the solar surface, the depth of which depends both on the type of mode and on its geometry. Some cavities are quite shallow, whereas others extend to the very center of the star. Since the frequencies of the modes are determined by the stratification and dynamics of the portion of the sun where their amplitudes are appreciable, accurate determination of the frequencies by observations affords remarkable ways of probing the inside of the star. During the past decade there has been significant progress in both observation and theory. The unravelling of information from many modes forms the basis of the subject called helioseismology. It promises study of the inside of the sun in sufficient detail to test the premises of stellar structure theory, and thus has implications for much of astrophysics. It can also provide crucial information about the temperature and mass distribution within the sun for testing issues in particle physics and theories of gravitation.

9:30

3aPA3. Physical acoustics at 500 GHz: 31 octaves above middle C. Humphrey J. Maris (Dept. of Phys., Brown Univ., Providence, RI 02912)

Experiments are described in which the techniques of picosecond optics are used to generate and detect ultrasonic waves at frequencies as high as 500 GHz. The method also provides exceptionally good spatial and temporal resolution, thereby making possible pulse-echo acoustic measurements on a broad range of nanostructures of current technical interest. Examples will be given of results that have been obtained by this approach, and possible future developments will be discussed.

10:00–10:15 Break

10:15

3aPA4. Synchronous picosecond sonoluminescence. Bradley P. Barber, Robert Hiller, Ritva Lofstedt, and Seth Putterman (Phys. Dept., Univ. of California, Los Angeles, CA 90024)

A strong sound field can (1) trap a bubble of gas at a velocity node, (2) maintain the bubble against diffusion, and (3) at sufficient intensity cause it to emit flashes of light that have intensities over 30 mW and widths less than 50 ps. The flashes are emitted in a clocklike fashion with a jitter that is also less than 50 ps. The spectral intensity of the sonoluminescence (SL)

increases into the ultraviolet. The mechanism for this effect is still unknown but light scattering experiments have resolved the bubble motion—radius versus time—on a scale of nanoseconds and indicate that the light is emitted by a supersonic collapse. The emitted intensity is a strong function of ambient temperature and varies by a factor of 200 between 1 and 40 C. Light scattering measurements of temperature-dependent SL therefore provide data that can critically test any proposed theory such as imploding shock waves [C. C. Wu and P. H. Roberts, *Phys. Rev. Lett.* (May 1993)]. [Work supported by the US DOE Division of Advanced Energy Projects, R. L. is supported by an AT&T fellowship.]

10:45

3aPA5. Evolution of the scanning microscopes both for imaging and lithography. C. F. Quate (Edward L. Ginzton Lab., Stanford Univ., Standard, CA 94205)

In recent times the atomic force microscope and other scanning probes have emerged as formidable instruments for imaging insulating surfaces. They can be operated in air, in liquid, and in vacuum and they have had an impact in the fields of electrochemistry and in cellular biology. Here, focus will be on solid surfaces where the AFM has been successfully used to image a variety of surfaces that are difficult to study with the optical and electron microscopes. Topography and microroughness of silicon wafers will be used to illustrate the utility of this instrument. Surface modification on the atomic scale has been demonstrated for a variety of systems. In a second application the force microscope has been combined with other forms of microscopes. The combination provides the operation with a simple means of controlling the spacing between tip and sample throughout the scanning cycle. This principle is illustrated with a description of the near-field optical microscope as combined with the force microscope. This combination allows one to scan with the tip space 5 nm from the sample.

11:15

3aPA6. Quantitative ultrasonic imaging for tissue characterization. James G. Miller, Samuel A. Wickline, Julio E. Perez, Benico Barzilai, Mark R. Holland, Scott M. Handley, and Burton E. Sobel (Dept. of Phys. and Cardiovascular Div., Washington Univ., St. Louis, MO 63130)

This presentation will illustrate relationships between physical acoustics and ultrasonic imaging for medical diagnosis. Ultrasonic tissue characterization is designed to complement two-dimensional echocardiography by providing information in addition to that derived from an assessment of tissue dimension and motion. The hypothesis is that indices based on (frequency-dependent) backscatter and attenuation can provide a noninvasive tool for the early diagnosis of diverse disease processes including ischemia and cardiomyopathy. Differentiation between mature infarct and acutely ischemic myocardium, which might benefit from reperfusion by angioplasty or thrombolysis, appears to be feasible on the basis of the frequency dependence of backscatter, which is lower in zones of infarct than in acutely ischemic or normal myocardium. Two-dimensional images are formed from scans made with the direction of propagation of ultrasound at varying angles relative to the local fiber orientation of the myocardium. Backscatter and attenuation vary substantially with the angle of insonification relative to myofibers. This research is designed to lay the groundwork for exploiting the anisotropy of myocardial ultrasonic properties to achieve improved understanding of cardiac mechanical properties (elasticity and compliance) in normal and diseased hearts. [Work supported by NIH HL40302, HL17646, HL42950.]

Contributed Papers

11:45

3aPA7. Some light emission features of single bubble sonoluminescence. Sean M. Cordry (Dept. of Phys., Univ. of Mississippi, University, MS 38677), Lawrence A. Crum, and Ronald A. Roy (University of Washington, Seattle, WA 98105)

Bubbles created via electrolysis were allowed to rise through water in a quiet acoustic levitation vessel. The sound field was then activated, forcing several bubbles to converge and coalesce near a pressure antinode. Light emission measurements were then taken with a Hamamatsu photomultiplier tube and single photon counter as a state of single bubble sonoluminescence (SBSL) evolved. The measurements reveal brief periods of fluctuations in light emission intensity followed by long periods of stable (i.e., nonfluctuating) emission. The time scales for the fluctuations are on the order of half a second. Previous measurements of light emission have indicated that SBSL exhibits remarkably long-lived, stable behavior [Gaitan *et al.*, *J. Acoust. Soc. Am.* **91**, 3166 (1992); B. P. Barber and S. J. Putterman, *Nature* **352**, 318 (1991)]. These measurements, however, imply a transient regime that the bubble must pass through while seeking a final position of stability. The highest light emission intensities are seen in the transient regime. These transient light emission should provide important information concerning the mechanisms through which SBSL develops its remarkable stability. Further, acoustic emissions from SBSL were observed to evolve from

broadband to narrow-band noise during the transient period. These light emissions and acoustic emissions will be presented and possible correlation discussed. [Work supported by ONR.]

12:00

3aPA8. Similitude in thermoacoustics. Jeffrey R. Olson and Gregory W. Swift (Condensed Matter and Thermal Physics Group, Los Alamos Natl. Lab., Los Alamos, NM 87545)

The principle of similitude has been applied to thermoacoustics. Using similitude reduces the number of variables necessary to describe an experiment, which can greatly reduce experimentation time. In addition, it provides insight into the building of models with identical performance characteristics. Such models would duplicate even unanticipated nonlinear behavior. This allows one to build test prototypes that operate at conditions which are more easily accessible than the desired final product. For example, a prototype that operates at modest pressures using heavy gas such as argon could model equipment that is intended to operate at high pressure using light gas such as helium. Similitude has been verified using a large thermoacoustic engine operating with helium, neon, and argon, and has been found to be obeyed almost perfectly, even for nonlinear effects whose origin is not understood. Therefore it is believed this principle can be applied to make meaningful predictions using simpler, cheaper test models.

Session 3aPP

Psychological and Physiological Acoustics: Cochlear Physiology, Hearing Impairment, and Measurement of Auditory Function

Jane Opie, Chair

Psychoacoustics Laboratory, Department of Speech and Hearing Science, Arizona State University, Tempe, Arizona 85287

Chair's Introduction—8:25

Contributed Papers

8:30

3aPP1. Direct evidence for automatic gain control in the cochlea. J. J. Zwislacki and M. Chatterjee (Inst. for Sensory Res. and Dept. of Bioeng. and Neurosci., Syracuse Univ., Syracuse, NY 13244-5290)

Compression of the input/output functions of auditory nerve fibers was ascribed to automatic gain control by Rose and his associates many years ago. It was ascribed to synaptic processes [C. D. Geisler and S. Greenberg, *J. Acoust. Soc. Am.* **80**, 1352-1363 (1986)]. But compression is known to take place in the cochlea, as has been documented at the basilar membrane and hair cell levels. By comparing measured waveform distortion and the size of the second harmonic to those predicted from the input/output functions of Hensen's cells, it is demonstrated that the cochlear compression is due to automatic gain control at least up to a sound pressure level of 70 dB in Mongolian gerbils. Strong compression has been found at levels as low as 20 dB SPL. Since the response magnitude of Hensen's cells has been shown by the present authors and others to be directly proportional to that of the outer hair cells, this demonstration should be applicable to the hair cells themselves. The almost complete lack of waveform distortion was present independent of frequency distance from CF and could not have been caused by spectral filtering. [Work supported by NIDCD Grant No. DC00074.]

8:45

3aPP2. Intracellular transfer functions in the apical cochlear turn: Implications for the pitch code. Monita Chatterjee and Jozef J. Zwislacki (Inst. for Sensory Res., Merrill Ln., Syracuse Univ., Syracuse, NY 13244-5290)

Intra- and extracellular ac transfer functions were recorded in the 500-Hz to 1-kHz region of the Mongolian gerbil cochlea. In agreement with previous results obtained in the 2-kHz location, the best frequency decreased with increasing SPL, the shift ranging from 1/2 to 1 oct. The phase transfer function also exhibited a shift from low to high SPLs amounting to about 180 deg. Damage to the preparation resulted in smaller responses, lowered EP, a reduction in the peak shift, as well as a disappearance of the phase shift at high SPLs. The normal transfer functions demonstrate a conspicuous high-frequency notch and secondary maximum at moderate and higher intensities. The finding that, in the range between 500 Hz and 2.5 kHz, the BF is intensity-dependent, indicates that the place of maximum excitation cannot be an adequate code for pitch. It has been suggested previously with respect to the 2-kHz location, that the one intensity-independent feature of the transfer functions is the high-frequency cutoff. This seems to be approximately true for the apical turn of the gerbil also. [Work supported by NIDCD.]

9:00

3aPP3. The influence of frequency resolution on the detection of spectral contrast by hearing-impaired listeners. Van Summers and

Marjorie R. Leek (Army Audiol. and Speech Ctr., Walter Reed Army Med. Ctr., Washington, DC 20307-5001)

Abnormal frequency resolution associated with sensorineural hearing impairment produces a smearing of spectral detail in the internal representation of complex acoustic stimuli. As a result, listeners with hearing loss may have difficulty locating spectral peaks that provide important cues to speech understanding. This study examined the relationship between frequency separation of peaks in a complex sound and the degree of spectral contrast preserved in normal and impaired auditory systems. Five hearing-impaired subjects (HI) and five normal-hearing subjects (NH) discriminated a flat-spectrum bandpass stimulus from a stimulus containing a sinusoidal ripple across its frequency range. The peak-to-valley amplitude (in dB) necessary for detection of the ripple was measured for ripple frequencies ranging from 1 to 9 cycles/oct. Auditory filter characteristics in notched-noise were measured at 1 and 3 kHz in order to assess the relationship between frequency resolution and the ability to detect spectral contrast in complex spectra. There were clear differences between groups in both auditory filter characteristics and spectral contrast detection. Auditory filters tended to be both broader and more asymmetric for HI listeners than for NH listeners. Mean ripple amplitudes at threshold were approximately 4 dB lower for the NH group than the HI group at all ripple frequencies. However, excitation patterns based on auditory filter characteristics and threshold stimuli specific to individual listeners exhibited levels of peak-to-valley contrast that were nearly identical across listeners. This suggests that increased ripple detection thresholds of the hearing-impaired listeners were due to spectral smearing resulting from impaired frequency resolution. [Work supported by NIDCD Grant No. DC00626.]

9:15

3aPP4. Formant transition discrimination at varying frequency regions among normal hearing young and elderly adults as a function of noise, frequency region, and transition direction. Jane F. MacNeil (Dept. of Psychol., Univ. of Calgary, 2500 University Dr., N. W., Calgary, AB T2N 1N4, Canada)

This study investigated the perceptual abilities of young (18-28 years) and elderly (65-75 years) adults to discriminate suprathreshold short duration dynamic frequency transitions. Influences of frequency region (1 to 3 kHz), transition direction (upward or downward gliding), offset frequencies (converging or diverging), and background condition (quiet or noise) were examined. Signals were 60-ms transitions classified according to whether or not: they diverged to differing offset frequencies, converged on a common terminal frequency; and had an upward or downward trajectory. Young adults displayed smaller just-noticeable differences (jnds) relative to the elderly, and the elderly showed greater effects for frequency region, noise, and transition direction. Signals centered at 3 kHz were considerably more difficult for the elderly to distinguish even though hearing sensitivity levels were comparable across both groups at this region. However, among both age

groups diverging signals produced smaller jnds than did converging signals. The results of the study showed that elderly subjects with comparable audiological profiles to young adults, or a mild sensorineural hearing loss, have greater difficulty distinguishing dynamic sounds, especially in background noise.

9:30

3aPP5. Stream segregation in normal-hearing and cochlear-impaired listeners. J. H. Grose, J. W. Hall, and L. Mendoza (Div. Otolaryngol./Head & Neck Surgery, Univ. North Carolina, Chapel Hill, NC 27599-7070)

Cochlear-impaired and normal-hearing listeners were compared on two tasks intended to assess auditory stream segregation. One task measured gap discrimination between two tones that were presented either in isolation or were embedded into a sequential stream of tones. For tone pairs distally spaced in frequency, performance was poor when the tones were presented in isolation but improved when they were drawn into separate auditory streams, presumably because temporal judgments could now be made within a stream. Both groups of listeners gave similar patterns of results although some differences were apparent. The second task was a melody recognition procedure [J. A. P. M. de Laat and R. Plomp, *J. Acoust. Soc. Am.* **78**, 1574–1577 (1985)] and involved identifying a 4-note target melody embedded between two competing melodies as a function of the proximity and temporal synchrony of the competing melodies. In both groups, marked individual differences were observed. Results from both tasks will be discussed in terms of the effects of cochlear impairment on auditory grouping skills. [Work supported by the NIDCD R01-DC01507.]

9:45

3aPP6. Modulation detection interference in subjects with a mild, high-frequency hearing loss. Jane M. Opie and Sid P. Bacon (Psychoacoust. Lab., Dept. of Speech and Hear. Sci., Arizona State Univ., Tempe, AZ 85287-1908)

Modulation detection interference (MDI) was measured in three normal-hearing subjects and in three subjects with mild, high-frequency sensorineural hearing impairment. Subjects detected 10-Hz amplitude modulation of a signal carrier in quiet and in the presence of a masker carrier that was unmodulated or amplitude modulated (depth of 100%) from 2 to 40 Hz. The carrier frequencies were 984 and 3952 Hz, either could serve as the masker, while the other served as the signal. Thus, for the hearing-impaired subjects, one carrier was in the region of hearing loss. In experiment 1, both the signal and masker were presented at equal sound pressure levels (SPLs). In experiment 2, the effect of sensation level (SL) was considered. In both experiments, there was generally little difference between the two groups when the signal carrier was in a region of normal hearing. However, the hearing-impaired subjects demonstrated more MDI than did the normal-hearing subjects when the signal was presented within the region of hearing loss. This suggests that even mildly hearing-impaired subjects may have difficulty processing complex, time-varying stimuli. Further, it appears that this may not be solely attributable to audibility but may be mediated by other factors. [Work supported by NIDCD.]

10:00–10:15 Break

10:15

3aPP7. Localization of sound sources in the median sagittal plane by listeners with high-frequency hearing loss. Timothy J. Vander Velde, Brad Rakerd (Dept. of Audiol. and Speech Sci., Michigan State Univ., East Lansing, MI 48824), and William Morris Hartmann (Michigan State Univ., East Lansing, MI 48824)

Sources in the median sagittal plane are localized on the basis of spectral cues. The elevation of sources toward the front is determined from cues at high frequency (7 kHz and above). The distinction between front, overhead, and back (FOB) potentially involves broadband

cues. One therefore expects that individuals with high-frequency hearing loss will fail at elevation tasks, though they might succeed at FOB tasks. Listeners with bilateral sensorineural hearing loss, moderate in the speech range and severe at high frequencies, were given source identification tasks in both elevation (three sources spanning 30 deg) and FOB geometries. The stimuli were white noise bursts, ranging in level from 42 to 90 dB SPL. A control experiment showed that the listeners could successfully identify sources in the left-overhead-right plane. Results for the sagittal plane showed that all listeners performed significantly less well than normal controls. In the FOB task, performance was well above chance, though listeners characteristically lacked either a front sensation or else a back sensation. In the elevation task, performance was, with a few exceptions, near chance for all listeners and all levels. [Work supported by the NIDCD, DC00181.]

10:30

3aPP8. Vowel discrimination after multichannel compression. E. William Yund and Thomas R. Crain (VA Med. Ctr., 150 Muir Rd., Martinez, CA 94553)

The effect of full-range multichannel compression (MCC) on vowel discrimination was studied in hearing-impaired and normal-hearing subjects. For normal-hearing subjects MCC was fit to four hypothetical flat losses, with thresholds ranging from 60 to 90 dB SPL, and one hypothetical sloped loss, with thresholds normal at 500 Hz and 90 dB SPL at 4 kHz. Each hearing-impaired subject was tested with the flat MCC systems as well as one fit to the subject's own hearing loss profile. Compression ratios varied from 1.75 to 7.00, in the flat MCC, and the number of channels varied from 2 to 31. Robinson-Huntington compression [C. E. Robinson and D. A. Huntington, *J. Acoust. Soc. Am.* **54**, 314 (1973)] had a 10-ms time window in all channels. Unprocessed stimuli and frequency-equalized linear amplification were control conditions. The expected deleterious effects of MCC on vowel discrimination are clear for the most severe compression systems, but preliminary results indicate that they fall off rapidly as the number of channels and compression ratios decrease. Complete results and their implications for the application of MCC in hearing aids will be discussed. [Work supported by Department of Veterans Affairs.]

10:45

3aPP9. The roles of parametric demodulation, rectifying demodulation, and coherent demodulation in human amplitude-modulated ultrasonic hearing. Shengke Zeng and Richard B. Beard (Biomed. Eng. and Sci. Inst., Drexel Univ., Philadelphia, PA 19104)

The frequency pattern of the human auditory sensation in response to amplitude-modulated (AM) ultrasonic stimulation is observed to be proportional to the frequency spectrum of the square of the modulation functions [Zeng and Beard, *J. Acoust. Soc. Am.* **93**, 2332 (A) (1993)]. In contrast, the frequency pattern of rectifying demodulation is proportional to the frequency spectrum of the absolute values of the modulation functions; and the frequency pattern of coherent demodulation is the spectrum of the modulation functions. The distinct discrepancies in frequency patterns between the auditory sensation in human ultrasonic hearing and the above two physical demodulations indicate that rectifying demodulation and coherent demodulation are irrelevant to human ultrasonic hearing. When the human subject's cochlear region is stimulated by an AM ultrasonic mode $(1 + \cos 2\pi Ft)\cos 2\pi ft$, the subject can sense the fundamental modulation frequency F and the weaker second harmonic $2F$, where f is the carrier frequency. The amplitude ratio of the intensity threshold for sensing the first harmonic to that for sensing the second harmonic is 0.37 at $F=1$ kHz, 0.57 at 250 Hz, and 0.68 at 3 kHz. Compared with the same ratio in parametric demodulation, which is 0.9, and in the demodulation caused by the quadratic property, which is 0.25, parametric demodulation is considered to have a joint influence in conjunction with the quadratic property on human AM ultrasonic hearing [Zeng, Ph.D. dissertation, Drexel Univ. (1993)]. [Work partially supported by Electro-Stim Corp.]

11:00

3aPP10. Effects of repetition rate, phase, and sex on the auditory brain-stem response in normal hearing subjects. T. K. Parthasarathy (Dept. of Speech Pathol. and Audiol., Southern Illinois Univ., Edwardsville, IL 62026 and Central Inst. for the Deaf, St. Louis, MO 63110)

The effects of click repetition rate and phase on wave I and V latency of the auditory brain-stem response (ABR) were evaluated in ten male and female normal hearing subjects. Clicks were presented monaurally via an insert earphone at 70 dB nHL, using either a condensation (C), a rarefaction (R), or an alternating (Alt) stimuli at two rates, 11.1 and 61.1/s. Females had shorter absolute (V) and interpeak latencies (I-V) than the males. These sex-related latency differences were phase independent. Furthermore, the stimulus phase effect on ABR latencies (I and V) were insignificant. Increasing the repetition rate produced greater wave I and V latency shifts for C and Alt clicks than the R clicks. Results seem to suggest significant rate-phase interaction effects on ABR latencies (I and V). However, the phase-sex and the rate-sex interaction effects on ABR latencies (I and V) were insignificant.

11:15

3aPP11. Effect of click-repetition rate on acoustic reflex growth. Vishakha W. Rawool (Commun. Disord. & Special Educ., Bloomsburg Univ., Bloomsburg, PA 17815)

Effects of click-repetition rate on acoustic reflex growth were studied in 11 young female subjects (total 22 ears). The probe tone frequency was 226 Hz and the intensity was 85 dB SPL. Following the determination of ipsilateral acoustic reflex thresholds in response to condensation clicks at repetition rate of 50, 100, 150, 200, 250 and 300/s, reflex amplitudes were measured at levels 5 and 10 dB above the thresholds. For each of the measurements data were obtained across three trials. In 10 of the ears evaluated, data were also obtained at 95 dB peSPL for each of the repetition rate that revealed that acoustic reflex amplitudes increase with increase in repetition rates. A significant effect of repetition rate and sensation level and a significant interaction was apparent for the amplitudes obtained at 5- and 10-dB sensation level (SL). For all the repetition rates 10 dB SL yielded higher amplitudes than those obtained at 5 dB SL except at the 50/s rate where increase in sensation level did not produce any significant increase in amplitudes. Detailed analyses will be presented and the results will be discussed with reference to rate integration in the acoustic reflex pathway. [Work supported by Bloomsburg University Grants for Research and Creative Projects and State System of Higher Education Minority Faculty Development Fund, Pennsylvania.]

11:30

3aPP12. Rate integration in click-evoked acoustic reflex thresholds. Vishakha W. Rawool (Commun. Disord. & Special Educ., Bloomsburg Univ., Bloomsburg, PA 17815)

Effects of click repetition rates on acoustic reflex thresholds were investigated in 11 female subjects (total 22 ears) within the age-range of 20 to 26 years. Acoustic reflexes were elicited in response to ipsilateral condensation clicks of 100- μ s duration at repetition rates of 50, 100, 150, 200, 250, and 300/s. The probe tone frequency was 226 Hz and the intensity was 85 dB SPL. Acoustic reflex thresholds were defined at the lowest intensity at which a minimum of 0.02-ml change in admittance was evident on at least two of three trials. The thresholds became significantly better with increase in the repetition rates. The average threshold advantage for the 300/s-click rate over the 50/s rate was 22.5 dB and that of the 100/s rate over the 50/s rate was 12.5 dB. Although such advantage with increase in stimulus repetition rates has been most commonly referred to as temporal integration, a more appropriate term for this phenomenon may be rate integration. Since the firing rates of auditory neurons are known to increase with increase in stimulus intensity, it is conceivable that stimuli presented at higher repetition rates are perceived as being louder. [Work supported by Bloomsburg University Grants for Research and Creative Projects and State System of Higher Education Minority Faculty Development Fund, Pennsylvania.]

11:45

3aPP13. Effects of vigilance on cortical processing. D. M. Daly (Box 210855, Dallas, TX 75211)

Normal persons significantly deprived of sleep show impaired vigilance with slowing of motor responses, errors in cognitive processing, and defects in recall. Pupillometry measures autonomic correlates of vigilance. Effects of sustained and of fluctuating vigilance on cortical processing using standard sets of synthetic acoustic stimuli are reported. [Daly *et al.*, *J. Neurophysiol.* **44**(1), 200-222 (1980)]. A 60-yr-old woman had lifelong difficulty awakening, and if not awakened might sleep 14 h continuously. A single 7 am 30-mg dose of methylphenidate aided arousal; once awake she could remain alert. In 2-h sessions of unmedicated continuous testing she classified stimuli consistently [$p(\chi^2 > G^2)0.0001$] and appropriately ($p0.001$, $re: 32$ normal adults). Her performance after 1 $\frac{1}{2}$ h was as consistent as the best controls. A 54-yr-old man had since adolescence slept less than 3 h per night (typically 2-5 pm). Unlike other reported philagrypnics, for 30 yr he has also napped for an hour in late morning. Despite diminished sleep requirements, he does not remain alert while awake. Performance with FM transients in CVs fluctuated widely ($p < 0.01$ to $p < 0.0001$), but remained consistent with relatively CF vowels ($p < 0.0001$). His performance improved temporarily with caffeinated (but not decaffeinated) coffee. These results are consistent with evidence that vigilance modulates cortical auditory processing; they suggest that like capacity for sleep, individual capacity to remain alert may also vary in a Gaussian distribution.

Session 3aSA

Structural Acoustics and Vibration: Inhomogeneous Structures and Active Control

Jerry H. Ginsberg, Cochair

School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332

Chris R. Fuller, Cochair

Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, Virginia 24061

Chair's Introduction—8:00

Invited Paper

8:05

3aSA1. Reciprocity theorem for oscillations of inhomogeneous elastic structures in a liquid—Modified reciprocity theorem and applications. Leonid M. Lyamshev (N. N. Andreev Acoustical Institute, Shvernik str. 4, 117036 Moscow, Russia)

The oscillatory vibrations of inhomogeneous elastic structures are considered. In particular, two problems involving plates and shells with attached ribs excited by forces and moments are discussed. The formulation uses the Green's function for the wave equation of sound propagation in a liquid, and the equations of motion of plates and shells coupled with those of flexural and torsional vibrations of ribs. An integral relationship connecting the solutions of two self-conjugated boundary problems of inhomogeneous plates and shells oscillations theory is obtained. This relationship is the reciprocity theorem that is a precise mathematical formulation of the well-known Rayleigh reciprocity principle for oscillating mechanical systems. If a liquid is moving and there is a shear flow outside or inside of a shell (plate) then this relationship is the modified reciprocity theorem for inhomogeneous elastic structures oscillations in a flow that is the generalization of the Rayleigh principle. Applications of the reciprocity theorems are discussed. Flexural waves reflection, scattering, and radiation by elastic ribs in plates are considered. Nonspecular reflections and resonance phenomena are observed. These are similar to nonspecular reflection, sound scattering, and radiation by elastic bodies in a liquid and are connected with interactions of flexural waves in a plate with flexural and rotational waves in ribs.

Contributed Papers

8:30

3aSA2. Approximate traveling wave solution for impulsively perturbed circular membrane. Alan Powell (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792)

It is instructive to transform the normal modes solution for a circular membrane (radius R) suddenly displaced ξ_0 at the boundary into an approximate purely periodic traveling waveform $\xi = \xi_0 \sum \times A_n \sqrt{r/R} \cos[(n - \frac{1}{4})\pi(r/R \pm ct/R) - \pi/4]$, where c = wave speed (although strictly good for $n \gg 1$, $r/R \ll 1$). The series evidently has period $T = 8R/c$. Individual mode periods $T/3, T/7, T/11, \dots$ account for the notable characteristic symmetry of the waveform about the center of the periods T and asymmetry about the quarter points. The $-\pi/2$ focal phase shift (finite Hilbert transform) and the simple sign reversal ($\pm\pi$ shift) at the boundary are readily deduced. The result also applies to ideal supersonic jets, $\xi \Rightarrow$ pressure perturbation, $ct \Rightarrow$ (axial distance) $\sqrt{M^2 - 1}$. For cell length (axial period) one may take $s_M = 8R\sqrt{M^2 - 1}$ for the series, or $s_P = \frac{8}{3}R\sqrt{M^2 - 1}$ for the first term. The two-dimensional case is unambiguous $s_M = 4$ (semiwidth) $\sqrt{M^2 - 1}$. Of course, the exact solution is not periodic at all, the major discrepancy being in first term approximation ($\frac{4}{3}$ vs 1.307...), so a better approximation is to transform just terms $n > 2$: the characteristic periodic waveform remains a dominant feature.

8:45

3aSA3. Localization and delocalization in the response of a beaded string. G. Maidanik and J. Dickey (CDNSWC, Bethesda, MD 20084-500)

In a complex structure in which the substructures are fairly periodic the phenomenon of localization (à la Anderson) and delocalization may arise when the periodicity is disturbed. In this paper, the localization and delocalization that may manifest in the response of a beaded (taut) string are discussed. It is argued that this phenomenon is related to the phenomenon of pass and stop bands that arise in a beaded string in which bays are fairly identical. A bay is the portion of a string between adjacent beads. When the identity of the bays is disturbed, localization and delocalization occur at frequencies that lie in pass and stop bands, respectively. The disturbance may be in the form of changing fractionally and randomly the spatial extents of the bays. Both localization and delocalization may play a significant role in the response behavior of periodic structures.

9:00

3aSA4. Vibration damping of large structures induced by attached small resonant structures. M. Strasberg and D. Feit (David Taylor Res. Ctr., Bethesda, MD 20084-5000)

It is well known that the observed vibratory behavior of large and complicated structures usually indicates much more damping than can be accounted for by the inherent dissipation in the structural material. Several investigators have pointed out that greatly increased damping may result from vibratory power dissipated by the numerous small resonant substructures usually attached to the large main structure; C. Soize called them the "structural fuzzy" [J. Acoust. Soc. Am. 92, 2365 (A) (1992)]. This paper presents a simplified procedure for estimating the effect of these substructures when they have many resonances at frequencies near the excitation frequency. In this case, the increased

damping of the large structure depends mainly on the total effective mass of all the substructures resonating within a frequency band (say a half-octave) centered at the excitation frequency, and is relatively independent of the specific amount of dissipation in the individual substructures. The effect of the substructures on the frequency response and vibration pattern of the main structure can also be accounted for in simple fashion. The influence of nonuniform spatial distributions of the attached substrates on the estimates is also discussed in terms of specific examples. [Work supported by ONR.]

9:15

3aSA5. Application of the Wiener-Hopf technique to the scattering of structural acoustic waves from discontinuities in fluid-loaded cylindrical shells. Steven L. Means (Graduate Program in Acoust., Appl. Sci. Bldg., Penn State Univ., University Park, PA 16801)

Several types of structural acoustic waves are known to propagate on fluid-loaded cylindrical shells. When propagating in the axial direction they can be described by a dispersion relation $D(k_x, \omega) = 0$. When these waves are incident on a shell discontinuity, here considered larger than the radius and smaller than the wavelength, energy is scattered back along the cylinder axis and into the surrounding medium. To characterize this phenomenon, the shell is considered to be locally reactive from $-\infty$ to 0, and rigid from 0 to ∞ along the x axis. A wave of known type is incident from $x = -\infty$ and excites a reflected wave and acoustic waves originating from the vicinity $x=0$. Pertinent asymptotic results are obtained including the reflection coefficient of the reflected wave and the far-field acoustic radiation pattern of the scattered wave. [Work supported by ONR and by the William E. Leonhard endowment to Pennsylvania State University.]

9:30

3aSA6. Accurate finite element methods for fluid-loaded plates. Karl Grosh (Div. of Appl. Mech., Dept. of Mech. Eng., Stanford Univ., Stanford, CA 94305-4040) and Peter M. Pinsky (Stanford Univ., Stanford, CA 94305-4020)

The design and analysis of new finite element methods for the steady-state response of fluid-loaded Reissner-Mindlin plates is presented. In this study, new methods are developed that are more accurate than standard finite element methods in their ability to represent wave propagation for coupled problems in structural acoustics. Generalized Galerkin least-squares (GGLS) finite element methods have been employed previously to enhance the accuracy of the finite element approximation for the uncoupled structural problem [K. Grosh and P. M. Pinsky, in *Proc. Second Int. Conf. Math. and Numer. Aspects of Wave Propagation*, edited by R. Kleinman *et al.* (June 1993)] and the uncoupled acoustic problem [I. Harari and T. J. R. Hughes, *Comput. Methods Appl. Mech. Eng.* **87** (1991)]. Complex wave-number dispersion analysis is used both to design the new methods and characterize their accuracy. Results comparing the finite element dispersion relations to the analytic dispersion relations for the fluid-loaded Reissner-Mindlin plate demonstrate the enhanced accuracy of these new GGLS methods over the standard Galerkin finite element implementation.

9:45

3aSA7. A spectral description for the effects of multiple point masses in fluid-loaded plates. Kenneth A. Cunefare, Jerry H. Ginsberg, and John Babish (The George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The wave-number-based formulation of the surface variational principle describes the surface pressure and displacement as a comparatively small set of interacting waves. It enables one to pose questions of parametric sensitivity from a global perspective. A two-dimensional problem of an elastic plate in an infinite baffle with pinned boundary conditions is considered. A series of line masses attached to the plate, at regular and irregular spacings is considered. The specific mass distribution is replaced in the SVP formulation by an arbitrary continuous distribution

along the length of the plate. The functional form of this distribution is described using a spectral Fourier series, whose ascending orders represent successive stages in refinement of the scale to which a model describes inertial effects. The excitation applied to the plate is taken as a concentrated harmonic force. With the excitation held fixed, the influence of each spectral component of inertial distribution on the surface response and radiated power are assessed. Evaluations carried out for a range of frequencies shed light on how small scale inertial heterogeneities can influence macroscopic radiation features.

10:00

3aSA8. The exterior acoustic modes of structures. Noelle Curry and Kenneth A. Cunefare (The George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The use of a modal-style approach for the analysis of the exterior radiation characteristics of structures continues to receive increasing attention. This approach generally seeks to find a set of orthogonal functions, or acoustic modes, that diagonalize a radiation operator in the exterior domain of an extended radiator. These acoustic modes are found through an eigenfunction/eigenvector analysis or singular value decomposition analysis of the radiation operator. The eigenvalue or singular value associated with a given mode is directly proportional to the radiation efficiency of that mode. Here, the frequency dependency of the acoustic modes of a baffled beam is examined. Further, the dependency of the radiation efficiencies and mode shapes on the number of degrees of freedom permitted in the derivation of the radiation operator is investigated. It is demonstrated that the accuracy of the acoustic modal representation depends on the number of degrees of freedom permitted in the derivation of the radiation operator. The most efficient acoustic modes are least sensitive to increasing degrees of freedom. The least efficient acoustic modes are most sensitive to changes in degrees of freedom. This behavior has significant impact on applications of the exterior acoustic modal approach that seek to exploit the least efficient modes.

10:15

3aSA9. Focusing of sound power flow for sound incident on a structure in water. Robert Hickling (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677)

Computation of sound-power flow for sound waves incident on a solid sphere in water shows various degrees of focusing along the axis and in the wake of the sphere depending on the material properties of the sphere. This contrasts with a rigid immovable sphere that has no sound-power flow inside and a shadow zone in the wake. Visualizations are presented that show the focusing effect as a function of frequency for a number of different materials.

10:30

3aSA10. A comparison of structural intensity formulations. Yong Zhang and J. Adin Mann, III (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50011)

Much past research concerning structural intensity has focused on the structural intensity formulation and measurement methods. For the structural intensity in a thin plate, different formulas were derived based on different assumptions. Pavic's structural intensity formula [G. Pavic, *J. Sound Vib.* **49**(2), 221-230 (1970)] was based on classical plate theory, which is described by Lagrange's equation of motion. A more general derivation was performed by Romano *et al.* [Proceedings of the International Congress on Intensity Techniques, CETIM, 137-142 (1990)], which was based on three-dimensional elasticity. In this paper, a structural intensity formula is derived by simplifying Romano's structural intensity formula using Mindlin's assumption for thin plate theory that considers the rotary inertia and shear effect. The differences between the formulations will be discussed. From plate velocity measurements for an aluminum plate at single frequencies, the structural intensity is calculated based on the above-mentioned intensity formulas. In

addition, the input mechanical force function is estimated. Results show that the input power from structural intensity and the input mechanical force match well with the experimental results.

10:45

3aSA11. The characteristics of the radiated noise from open grid and bascule bridges. J. M. Cuschieri (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

When vehicles travel across the grid section of an open grid or bascule bridge, noise is generated which for most type of grid configurations has a very strong tonal characteristic in the frequency range between 150 and 250 Hz. To investigate the mechanism by which this noise is generated, and identify possible noise mitigation procedures, a series of measurements of both sound and vibration levels have been performed on 12 bascule bridges with different types of grid. The grid types can generally be classified into three configurations, two of which are of a rectangular or square shape, while the third configuration has diagonal members and is generally referred to as four-way grid. For the rectangular-shaped grids, both the vibration and the sound spectra exhibited a dominant peak at a frequency that scaled with the speed of the vehicle and a typical length of the grid spacing. For the four-way grid, the spectrum did not show a very strong peak, but was more broadly distributed in frequency. The general overall sound level was not much different from one bridge grid type to another. The grid is an open structure and its radiation efficiency in the 200-Hz region is very low. This makes the tire a potentially significant source of noise. [Work sponsored by FDOT.]

11:00

3aSA12. Active control of sound power using acoustic basis functions. Koorosh Naghshineh (Acoustics and Radar Technol. Lab., SRI International, 333 Ravenswood Ave., Menlo Park, CA 94025) and Gary H. Koopmann (Penn State Univ., University Park, PA 16802)

An improved method of active structural acoustics control is presented that is based on the minimization of the total power radiated from any structure expressed in terms of a truncated series sum. Each term of this sum is related to the coupling between the orthogonal eigenvectors of the radiation impedance matrix (referred to as "basis functions") and the structural surface velocity vector. The basis functions act as surface velocity filters. These acoustic basis functions are found to be weak functions of frequency but their corresponding weighting coefficients increase monotonically with frequency. The minimization of the radiated power is shown to result in a structural surface velocity vector that couples poorly to those acoustic basis functions that account for high efficiency sound radiation. This strategy is demonstrated numerically for a clamped-clamped baffled beam in air. An unexpected benefit of the control strategy described is that it provides a rational procedure for selecting the number and placement of actuators and sensors on a structure for effective control. This development is significant since this procedure does not require *a priori* knowledge of the dynamics of the structure.

11:15

3aSA13. A regression-based approach for simulating feedforward active noise control, with regression diagnostics for assessing the impact of measurement noise. C. E. Ruckman (Acoust. Technol. Dept., David Taylor Model Basin, Bethesda, MD 20084-5000) and Chris R. Fuller (Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

Current numerical techniques for simulating feedforward active noise control in the frequency domain are mathematically equivalent to the techniques required for linear least-squares regression. Measurement error in the control system error sensors plays a role analogous to that of observation error in a statistical regression. In the statistics literature, regressions are always accompanied by regression diagnostics, i.e., com-

puted statistics that help assess the impact of observation error. However, discussions in the acoustics literature typically fail to address this issue. The present work models feedforward active control as a regression of complex-valued variables, and shows how to apply two basic regression diagnostics: the *F*-test and the *t*-test. Also discussed are the physical significance of the "cost function" being minimized by the regression, and the assumptions that must be made regarding the variance of the measurement error. The techniques are demonstrated by numerically simulating a simple system in which radiation from an axisymmetric cylindrical shell is controlled by oscillating forces applied on the shell surface. [Work supported by David Taylor Model Basin and ONR.]

11:30

3aSA14. Feedforward control with the higher-harmonic, time-averaged, gradient (H-TAG) descent algorithm. Dorene Kewley, Robert L. Clark (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708-0300), and Steve C. Southward (Lord Corp., Cary, NC 27511)

A novel adaptive filter structure has been proposed for the control of systems characterized by higher harmonic response. The control approach has been designated the higher-harmonic time-averaged, gradient (H-TAG) descent algorithm. In the H-TAG algorithm, a single frequency reference input is all that is required to implement the controller. The remaining harmonics are generated internally based upon simple trigonometric relationships. In addition, the algorithm is ideal for structures characterized by slowly time-varying parameters since no system identification is required to implement the algorithm. Results from simulations and experiments indicate that the proposed approach offers a unique method of achieving significant levels of attenuation in the harmonic vibration or acoustic response of structures.

11:45

3aSA15. Active structural acoustic control with broadband disturbances and real time structural wave-number sensing. Julien P. Maillard and Chris R. Fuller (Vib. and Acoust. Labs., Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061-0238)

The present work gives further developments and experimental testing of a new time domain structural sensing technique for predicting wave-number information and acoustic radiation from vibrating structures. Most structure-borne active sound control approaches now tend to eliminate the use of microphones located in the far field by developing sensors directly mounted on the structure. In order to reduce the control authority and complexity required to minimize sound radiation, these sensors should be designed to provide error information that is solely related to the radiating part of the structural vibrations, i.e., the supersonic wave-number components. The approach discussed in this paper is based on estimating supersonic wave-number components coupled to acoustic radiation in prescribed directions. The spatial wave-number transform is performed in real time using a set of point structural sensors with an array of filters and associated signal processing. Experimental results on planar radiators show that only a few point sensors are required to provide accurate radiation information over a broad frequency range. The use of the sensing approach is then experimentally demonstrated in the time domain LMS active control of broadband sound radiated from a vibrating plate. [Work supported by ONR.]

12:00

3aSA16. Design of feedforward ASAC systems by eigenfunction assignment. Ricardo A. Burdisso and Chris R. Fuller (Vib. and Acoust. Labs., Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061-0238)

The traditional design approach of feedforward control systems involves the selection of number and location of the actuators and sensors based on some physical understanding of the system. This empirical

methodology yields satisfactory results for simple structures and sinusoidal inputs. However, such an heuristic approach can easily result in an inefficient control system with unnecessary large number of control channels for complex structures and more realistic disturbances. In this work an efficient formulation is presented for the optimum design of actuators and sensors for structurally radiated sound reduction. The technique is based on the modification of the eigenstructure such that the system responds with the weakest set of modal radiators. The technique is applicable to both narrowband and broadband excitations. The formulation is demonstrated for controlling the odd-odd modes of a simply supported plate driven by a point force located at the center of the plate. The radiation due to the first three odd-odd modes is reduced with a single-input, single-output (SISO) controller. The control actuator and error sensor are implemented with piezoelectric (PZT) ceramics and polyvinylidene fluoride (PVDF) films, respectively. It is shown that the optimum actuator and sensor configuration yields excellent global sound reduction. [Work supported by ONR.]

12:15

3aSA17. Adaptive control of bending wave intensity in a finite beam. David C. Swanson, Cassandra Gentry, Sabih I. Hayek, and Scott D.

Sommerfeldt (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802)

Adaptive control of the bending wave field in a finite beam is simulated using a classical Euler-Bernoulli analytical model and a dual-actuator filtered-x feedforward controller. Three different intensity-based error signal strategies are compared for multiple locations on the beam in and out of the near field of the actuators. The goal is to actively minimize the propagating bending wave power where the five-element error sensor array may be located in the near field of the actuators. Since the error array must be effective in the near field, the idea of simultaneously minimizing all five accelerometer signals was rejected. Results from using an instantaneous intensity error estimate were ineffective due to the nonlinearity in the adaptive control error gradient. Results from minimizing the intensity (rather than intensity-squared) were acceptable and consistent with the experimental results of Sommerfeldt [Sommerfeldt *et al.*, *J. Acoust. Soc. Am.* 93, 2370 (A) (1993)]. A new error method based on an intensity transfer function plant also produced acceptable simulation results. [Work supported by the GEM Fellowship Program.]

WEDNESDAY MORNING, 6 OCTOBER 1993

MAJESTIC BALLROOM, 9:00 TO 11:40 A.M.

Session 3aSP

Speech Communication: Tribute to Franklin S. Cooper

Katherine S. Harris, Chair

Haskins Laboratories, 270 Crown Street, New Haven, Connecticut 06511

Chair's Introduction—9:00

Invited Papers

9:05

3aSP1. Technology for people. Mark Haggard (MRC Inst. of Hear. Res., University Park, Nottingham NG7 2RD, UK)

Successful application of science and technology to improve human welfare, as seen in the career of Frank Cooper, requires a sensitive awareness of human needs and limitations. This talk illustrates how the drive for scientific understanding has interacted with concern for human welfare in producing two types of instrument (auditory warnings and auditory diagnostic or screening devices) in which the Medical Research Council of the UK has made major investments in recent years. Successful development requires a good grasp of engineering principles, well-judged investment in enabling technologies, correct description of the need to be met, and evaluation of the real life usefulness of the devices.

9:30

3aSP2. Reading, reading machines, and communications research. Alvin M. Liberman (Haskins Lab., 270 Crown St., New Haven, CT 06511)

What knowledge or habit of mind must would-be readers command that mastery of speech will not have taught them? Surely, that is the first question one must answer if one is to understand the reading process and the ills that attend it. Yet the question could hardly be accommodated, let alone answered, within the theory of speech that was almost universally accepted when Frank Cooper entered the field. It therefore counts as a major achievement of his research that others were able to gain from it a critical insight into the relation between the biologically primary processes of speech and the biologically secondary processes of reading, and thus to see more clearly the difficulties that beset the progressions from the one to the other.

9:55

3aSP3. Speech synthesis by rule, text-to-speech, and aids for persons with disabilities. Hector Raul Javkin (Speech Technol. Lab., Panasonic Technol., Inc., 3888 State St., Santa Barbara, CA 93105)

Speech synthesis by rule and its subsequent development into text-to-speech have progressed into both science and technology for use by the general population, particularly by persons with disabilities. This paper briefly reviews the advances in speech synthesis by rule and text-to-speech and then focuses on a new development—the use of text-to-speech in a speech training system for deaf children. The method uses the acoustic parameters that a text-to-speech system supplies to its formant synthesizer and

converts them to pseudoarticulatory parameters equivalent to parameters measured from instruments monitoring the child's production. For example, the relative frequencies of the nasal pole and nasal zero are converted to a "nasalization index" equivalent to the output of a nasal sensor. The method enables a student to type any utterance she/he wants to learn and see a representation of the articulation of that utterance that corresponds to the feedback received from instruments. Preliminary testing of the use of the method will be reported.

10:20

3aSP4. Design of auditory prostheses to aid speech communication. Sigfrid D. Soli and Robert V. Shannon (House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 91024)

Research on the treatment of hearing impairment and deafness has, in recent years, increasingly focused on auditory prostheses that can aid speech communication in both quiet and noise. For these prostheses to be most effective, several research goals must be achieved. First, the residual capacity of the impaired auditory system must be determined. Next, the essential acoustic information for speech recognition must be characterized and re-coded to exploit the residual auditory capacity. Finally, algorithms and circuits that can extract and process this information are required. Efforts toward these goals in hearing aids and in cochlear and brainstem implants will be described. The binaural directional hearing capacity of individuals with sensorineural hearing impairment has defined the main focus of this hearing aid research because of its potential to improve speech communication in noise. In this implant research, the channel capacity and channel interactions of the electrically stimulated auditory system have received major attention because of the effects of these interactions on speech coding for multi-electrode implants. Methods for characterizing the residual capacity of an individual, techniques for using residual capacity to aid speech communication, and results from perceptual tests obtained with real-time, laboratory-based processors will be discussed.

10:45

3aSP5. Imaging techniques for the larynx and vocal tract. Hajime Hirose (Faculty of Medicine, Kitasato Univ., 1-15-1 Kitasato, Sagami-hara, 228 Japan)

A recently developed method of digitally imaging vocal fold vibration was applied for the analysis of the nature of diplophonia, which is defined as the simultaneous production by the voice of two separate tones. In the present system, a specially designed tele-endoscope is attached to a single-lens reflex camera that houses a solid-state sensor at the position of the film plate. When the shutter is released, an image scan is made under computer control, and image signals are stored in the image memory. The image can be reproduced and displayed for later analysis. In the case of diplophonia, it was revealed that there is an asymmetry in the vibratory frequency of the left and right vocal folds associated with quasiperiodic variation in the speech waveform. As for the imaging technique for the vocal tract, a preliminary result of observation of the inner structure of the tongue using tagging snapshot MRI will be presented.

Contributed Papers

11:10

3aSP6. Tracking the gliding tongue and lips: Articulatory undershoot or perceptual overshoot or ...? Leigh Lisker (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695 and Univ. of Pennsylvania, Philadelphia, PA 19104)

At the last ASA meeting it was shown that for a set of synthetic vowel-glide-vowel sequences varying in F_2 trajectories a range of frequencies can equally well serve as mid segments in patterns identified as English /iwi/ and /uyu/. Many of these mid segments, when lengthed and presented in isolation, resemble the high front rounded vowel [y], so that when short and intervocalic one might expect them to be heard as the glide [ɥ], at least by phonetically trained listeners. However, perceptual data suggest that intended [ɥ] in /i—/i/ is heard as /w/, while in /u—/u/ it is /y/. Assuming the syllabifications /i\$wi/ and /u\$yu/, one might then expect an initial [ɥ] to be heard as /w/ in [ɥi] and as /y/ in [ɥu]. Test data indicate otherwise. Unlike /iwi/ and /uyu/, /wi/ and /yu/ show no overlap of F_2 values in their initial steady-state segments—there is instead a range of F_2 values for which listeners report both *not*-/wi/ and *not*-/yu/. Thus for English /w,y/ the steady-state and transitional intervals have different perceptual weights initially and medially. [Work supported by NIH Grant No. HD-01994 to Haskins Laboratories.]

11:25

3aSP7. The stability of a temporal distinction: Vowel length in Thai. Arthur S. Abramson (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695 and Univ. of Connecticut, Storrs, CT 06269-1145)

This is part of a larger effort to test the robustness over speaking styles of acoustic properties that underlie phonological contrasts. So much of the knowledge comes from the study of citation forms that one might wonder whether in running speech the presence of much top-down information and other contextual cues does not lead to the impoverishment of some of the bottom-up phonetic information. A good candidate for possible instability is relative duration as a means of distinguishing phonemes. The Thai language, in both production and perception, uses relative duration to distinguish its "short" and "long" vowels. In this study vowel durations were measured under three conditions. (1) Eight minimal pairs of words were recorded in carrier sentences by four native speakers at two rates, normal and fast. (2) Casual unrehearsed conversations were recorded by two couples. (3) Many words and short expressions taken from each person's part of a conversation were read aloud by that person. The data show that the quantity distinction in Thai is rather stable. The overlap in the pooled data from running speech largely disappears once certain contextual factors are taken into account. [Work supported by NIH Grant HD-01994 to Haskins Labs.]

Session 3aUW

Underwater Acoustics: Shallow Water Noise I

Kenneth E. Gilbert, Chair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Chair's Introduction—8:00

Invited Papers

8:05

3aUW1. Shallow-water noise—A review. Robert J. Urick (11701 Berwick Rd., Silver Spring, MD 20904)

Like most aspects of underwater sound, the noise in the sea received its first quantitative attention during World War II, when a team under the venerable acoustician V. O. Knudsen made measurements in the shallow coastal waters off Southern California and Florida—ostensibly for the purposes of acoustic mines and harbor protection sonars. At that time too, the pernicious snapping shrimp of the warm shallow tropic waters of the Pacific were investigated. One of the principal features of shallow water noise is its variability—from place to place and time to time. Yet it may be said that there are only three major noise sources in shallow water: ships and other man-made activities at moderately close ranges, biologics, and the wind or waves. Strangely, when the first two of these are absent, the noise levels in shallow water at frequencies above 1 kHz or so are the same as in deep water; the so-called “Knudsen” curves have long been used in deep water. At lower frequencies, many shallow locations are often more quiet than deep water because of the absence of deep-going favorable transmission paths. When environmental conditions can be estimated, one's knowledge of shallow-water noise should be good enough to permit a prediction of the expectable noise level at an arbitrary location—at least as good as one's prediction capability of transmission loss out to useful ranges. [Work supported by the Applied Research Laboratory, The Pennsylvania State University, and the Office of Naval Research.]

8:30

3aUW2. Man-made noise in shallow water. Stephen K. Mitchell (Appl. Res. Labs., University of Texas, Austin, TX 78713)

In many important shallow water areas, merchant shipping, fishing, or seismic exploration activities are a prominent feature. In such regions, the ambient noise levels in the frequency range from approximately 10 Hz to 1 kHz are generally dominated by sounds related to these activities. Ambient noise levels are generally much higher than those typical of deep ocean regions; omnidirectional and beam levels show considerable temporal variability as noise sources move and change their activities. Noise data from measurements in shallow water regions will be summarized. In addition, the interaction between source characteristics, shipping densities and traffic patterns, propagation characteristics, and ambient noise statistics will be discussed. [This work is supported by the Office of Naval Research, Code 234, under block RL3B.]

8:55

3aUW3. Biological sources of noise in coastal waters. Peter L. Tyack and Terrance Howald (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Underwater biological sounds are transient events. Durations range from microseconds to tens of seconds, frequencies from < 10 Hz to > 100 kHz, and source pressure levels up to > 220 dB *re*: 1 μ Pa at 1 m. While each sound is produced by one individual, many sounds occur in choruses that may involve thousands of simultaneous sources. Some crustaceans produce loud impulsive noise in the 2–20 kHz range. These animals are common enough in many shallow areas to dominate the ambient near 10 kHz. Evening fish choruses seasonally dominate the ambient in many shallow waters in the 100–1000 Hz frequency range. While whales and dolphins are less numerous than the animals listed above, many species produce sounds of sufficient intensity to affect the ambient. For example, during their winter breeding season fin whales produce series of 20-Hz pulses lasting 1 s at source levels of 180 dB *re*: 1 μ Pa at 1 m. Dolphin echolocation clicks may reach source pressure levels > 220 dB with spectral peaks 50–150 kHz. Dolphins are also skilled mimics of manmade sounds, which may lead to unpredicted interference.

9:20

3aUW4. Acoustic radiation from breaking waves. Li Ding and David M. Farmer (Inst. of Ocean Sci., P.O. Box 6000, Sidney, BC V8L 4B2, Canada)

Observations of the sound from breaking surface waves using a small hydrophone array illustrate the temporal and spatial characteristics of this sound source and its dependence on the surface wave field. Such observations provide a source description that might have application in ambient noise model development. By tracking the propagation speed of the breaking event its wave scale is inferred; other measured properties include breaking length, duration, and spatial separation. Analysis indicates that the dependence of breaking probability on the fourth moment of the wave spectrum is consistent with a linear model. The measured speeds of breaking events imply that their scale is less than the dominant wind-wave scale. Group structure of wave

breaking, in which there is a tendency for waves to break repetitively at twice the wave period and spaced one wavelength apart, also occurs but is more readily identifiable at lower wind speeds.

9:45–10:00 Break

Contributed Papers

10:00

3aUW5. The source pressure field arising from wave-wave interactions in shallow-water environments. A. C. Kibblewhite and C. Y. Wu (Dept. of Phys., Univ. of Auckland, Private Bag, Auckland, New Zealand)

Theoretical treatment of the noise source induced by wave-wave interactions have been traditionally based upon the assumption of an infinitely deep ocean. Growing interest in the application of the wave-induced pressure field as an acoustic source in the determination of the sub-bottom geoacoustic structure in shallow-water areas calls for a study of the influence of water depth on the physical processes involved. This paper presents some results based on such an extension. It shows how the depth-dependent surface-wave dispersion relation and porosity in the upper layers effect the spectral properties of the resulting ULF noise field in shallow-water environments. [Work supported by ONR.]

10:15

3aUW6. Measurements of the azimuthal distributions of very low-frequency noise with high resolution in a shallow water site in the Mediterranean. Bruce K. Newhall (Appl. Phys. Lab., Johns Hopkins Univ., Johns Hopkins Rd., Laurel, MD 20723-6099)

An extremely long horizontal towed array was used to perform high-resolution noise measurements at very low frequencies. The directional measurements in one shallow site are compared with more extensive measurements in deep water. The very low-frequency noise at all sites is highly anisotropic due to the high azimuthal resolution of the discrete merchant shipping sources that dominate underwater noise at these frequencies. The deep water measurements were performed at sites in three oceans, covering an extreme range of shipping density conditions, however, the shapes of the azimuthal distributions of the noise levels are relatively insensitive to shipping density. The shallow water noise exhibits azimuthal spreading not apparent in the deep water. From the loudest sources this spreading may partly obscure the spatially quiet regions that are otherwise present in directions between loud ships. Although they have different shape parameters, both the shallow and deep water data fall between Weibull and gamma distributions, and are well fit by generalized gamma distributions. The effects of multipath fading are diminished by the spatial averaging across the long aperture. Low-resolution noise distributions, obtained from subapertures of the array, are dominated more by multipath fading, and approach gamma distributions.

10:30

3aUW7. Modeling of echoes from extended targets in shallow water. T. Baus, M. Rosario, W. Chang, and R. Deavenport (Naval Undersea Warfare Ctr., New London, CT 06320)

Echoes are generated and compared for extended targets in a shallow water environment using the total target-medium broadband frequency response function and the frequency representation of the transmitted signal. The two-way medium response is calculated using a hybrid SAFARI/generic sonar model (GSM) environmental model. The target frequency response is found using a model of distributed targets and elastic spheres. The combination of the target and medium frequency responses gives the full frequency response, or form function, for the target, medium system. The product of the total form function with the frequency spectrum of the incident signal is computed and on inverse FFT is performed to predict the received signal, or echo, from the object. [Work supported by ONR Code 231.]

10:45

3aUW8. Model calculation for deviation of ambient noise statistics from signal statistics in shallow water. Jacob George (Naval Res. Lab., Code 7176, Stennis Space Center, MS 39529)

It has been reported that cumulative distribution functions (CDFs) of a signal in shallow water obey normal distributions, while those of ambient noise deviate from this, in both cases independent of depth [J. George, *J. Acoust. Soc. Am.* **92**, 2304 A (1992)]. One cause of intensity fluctuations is internal waves, studied by path integral and moment methods in deep water [B. J. Uscinski *et al.*, *J. Acoust. Soc. Am.* **74**, 1474 (1983)]. In shallow water, internal waves have been shown to behave as solutions [J. Zhou and X. Zhang, *J. Acoust. Soc. Am.* **90**, 2042 (1991)]. For the time spans considered in the CDFs mentioned above (25–320 s) intensity fluctuations caused by surface waves cannot be ignored. Results of a model calculation designed to contrast the effects of a stable signal source below the surface with the effects of ambient noise sources moving with the surface will be presented. [Work supported by ONR.]

11:00

3aUW9. A modified complex starting field for shipping traffic noise modeling. Christopher J. Burkhalter (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529-5004)

In current ambient noise modeling efforts, commercial shipping is considered a major contributor to overall noise levels and is modeled accordingly. The propagation of shipping noise from the many sources to the receiver is effected by use of any of a number of high-fidelity acoustic models. Almost all of these models require an initial condition in the form of a vertical pressure field that is stepped out numerically in range. This starting field usually takes one of several forms: a simple or modified Gaussian distribution around a source depth, a normal mode starter, a simple image (acoustic dipole) starter, or a Pade-approximated parabolic equation starting field. The implications of a near surface source are discussed, as well as effects of low- versus high-angle propagation models and the effects of complex reflection from the necessarily turbulent surface. Additionally, the use of the Gaussian distribution as a sufficient starting approximate is analyzed for long-range, deep-water results. [Work supported by ONR and NRL.]

11:15

3aUW10. Description of a noise model for shallow water: RANDI-III. J. Ernest Breeding, Jr. (Naval Res. Lab., Stennis Space Center, MS 39529-5004), Marshall Bradley, Melanie Hebert, and Walt McBride (Planning Systems, Slidell, LA 70458)

Version III of the research ambient noise directionality (RANDI-III) noise model is described and illustrated. The objective of the model is to make predictions of the ambient noise at low to mid frequencies (10 to several hundred Hz) for highly variable environmental conditions typical of shallow water and coastal areas. The receiver can be either a horizontal or vertical line array. Shipping and local wind are assumed to be the dominant sources of noise. Ships are treated as discrete sources of noise. The ship noise is propagated to the elements of the receiver using a wide-angle finite element parabolic equation. The contributions from all ships are added coherently. The environmental and shipping information are automatically extracted from several databases. [Research supported by The Office of Naval Research, Program Element 62435N with technical management provided by the Naval Research Laboratory, Stennis Space Center, MS.]

11:30

3aUW11. Ambient noise predictions from deep to shallow water. J. Ernest Breeding, Jr. (Naval Res. Lab., Stennis Space Center, MS 39529-5004), Marshall Bradley, and Melanie Hebert (Planning Systems, Inc., Slidell, LA 70458)

The RANDI-III noise model is used to make predictions of the omni noise level, horizontal noise field, and the vertical noise field for real world examples of broad and narrow transects from deep to shallow water. For comparison, some noise predictions are also made with the ANDES-II noise model. This is of interest since these two noise models are fundamentally different in several ways, including the methods used to propagate noise and the approaches used to treat ship sources. It is found that at the deeper water portions of the transects, the RANDI-III and ANDES-II predictions of the omni noise levels and the horizontal noise fields can be similar. However, the RANDI-III and ANDES-II predictions of the vertical noise field at these same water depths are quite different. At the shallow water sites the predictions of noise are not possible with ANDES-II. Examples illustrating the differences and similarities of the RANDI-III and ANDES-II predictions, as well as the RANDI-III predicted variations in the noise along the transects from deep to shallow water will be presented. [Research supported by The Office of Naval Research, Program Element 62435N with technical

management provided by the Naval Research Laboratory, Stennis Space Center, MS.]

11:45

3aUW12. A ray-model theory on spatial correlation and vertical directivity of surface-generated ambient noise in shallow water. Renhe Zhang, Yi He, Baixian Zhu, and Dinghua Guan (State Key Lab. of Acoust., Chinese Acad. of Sci., Beijing 100080, People's Republic of China)

The ambient noise in shallow water is more complicated than that in the deep sea, and it has greater geographical and seasonal variations since there is a strong bottom interaction. A theoretical model of ambient sea noise including surface-noise sources and a stratified medium ocean is discussed. The noise sources are assumed to be statistically independent directional point sources distributed over the sea surface. The effects of the surface-source feature, velocity profile, and bottom property on ambient noise are studied. In general, the ray theory is suitable for calculating the contribution of near-field sources, whereas the normal-mode theory is appropriate for far-field sources. In this paper, a ray-mode theory suitable for near- and far-field sources is developed. The concise formulas for calculating the intensities, spatial correlation, and vertical directivity of ambient noise are present and some numerical simulations are given.

WEDNESDAY MORNING, 6 OCTOBER 1993

BILTMORE ROOM, 9:00 A.M.

Meeting of Accredited Standards Committee S2 on Mechanical Vibration and Shock

to be held jointly with the

U.S. Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock

S. I. Hayek, Chairman S2

Applied Research Laboratory, Penn State University, P.O. Box 30, State College, Pennsylvania 16801

D. F. Muster, Chairman, U.S. Technical Advisory Group (TAG) for ISO/TC 108, Mechanical Vibration and Shock
4615 O'Meara Drive, Houston, Texas 77035

Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees) including a report on the current activities of ISO/TC 108, and plans for the next meeting.

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

Session 3pAB

Animal Bioacoustics: Acoustical Monitoring of Animal Populations

John R. Potter, Chair

Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, California 92093-0238

Chair's Introduction—12:55

Contributed Papers

1:00

3pAB1. Tracking zooplankton at sea with a three-dimensional acoustical imaging system. Duncan McGehee and Jules S. Jaffe (Marine Phys. Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0205)

The Acoustical Imaging Group in the Marine Physical Laboratory, Scripps Institution of Oceanography, has developed a real time, high-resolution three-dimensional acoustical imaging system for use in the ocean. The system, called Fish TV (FTV) is designed for the observation of macrozooplankton and small fish. FTV can synoptically examine a 16° by 16° volume of seawater with a 1-Hz frame rate. The sonar has recently been used at sea in two configurations. In the first experiment, the sonar was mounted 40 m deep, looking laterally from the research platform R/P FLIP. A video camera provided ground-truthing. In the second experiment, the sonar was attached to a Phantom DS4 remotely operated vehicle and deployed, again with video ground-truthing. Results indicate that the system can be used both to track animals in three dimensions and to estimate their density.

1:15

3pAB2. Application and comparison of neural nets for marine mammal call classification. John R. Potter (MPL 0238, Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238) and David Mellinger (Cornell Laboratory of Ornithology, Ithaca, NY 14882)

Recent work has successfully applied a linear matched filter to calls made by Bowhead whales recorded off the coast of Alaska in frequency-time (spectrogram space) to detect and classify marine mammal calls. This method relies on an empirical matrix weighting for the matched-filter coefficients. A neural net, trained on spectrogram estimates as the feature vector space, offers two advantages over this approach: (a) the equivalent weighting matrix is determined by training and may coverage to a more optimal solution and (b) the response of a neural net is nonlinear and can embody more sophisticated responses. A simple three-layer feedforward neural net is ideally suited to this application and has been implemented on 204 calls, of which 163 were used for training and 31 kept as "unseen" test data. The neural net was configured to identify both whale calls and other mammal interference. The success rate including failures in both estimates on training data was 88%. The combined false-positive and false-negative whale detection errors on unseen data was only 7%, which compares very favorably with any other known method. The neural net approach is compared with the matched filter and the role of the hidden neurons and equivalent weighting matrix are discussed. [Work supported by the Office of Naval Research.]

1:30

3pAB3. Fortuitous underwater acoustic measurements of baleen whale vocalizations by large aperture arrays. G. L. D'Spain, J. A. Hildebrand, M. A. McDonald, and W. S. Hodgkiss (Marine Phys.

Lab., Scripps Inst. of Oceanogr., San Diego, CA 92152-6400)

Over the past several years, the Marine Physical Laboratory has designed, built, and deployed innovative, large-aperture sensor systems in order to measure the low-frequency properties of the ocean acoustic field. In many of these experiments, baleen whale vocalizations have been a significant feature of the infrasonic sound field. For example, finback whale (*Balaenoptera physalus*) sounds have been recorded by a 120-element, 900-m, vertical line array of hydrophones. Finback whale sounds were also recorded by a 12-element, 165-m, vertical array of "TRIFAR" elements (i.e., each element records simultaneously the three components of acoustic particle velocity and pressure). In addition, blue whale (*Balaenoptera musculus*) signals were the overwhelming feature of a data set recorded by some neutrally buoyant, freely drifting, acoustic particle motion/acoustic pressure sensors. Finally, new-design ocean bottom seismometers (OBS) have recorded both finback and blue whale signals. Where possible, these measurements are used to estimate both the position and the actual time signature of the source by removing the effects of multipath arrivals.

1:45

3pAB4. Detection of salt-marsh mosquito swarms in remote mangrove swamps. Richard W. Mankin (Agric. Res. Service, Insect Attractants, Behavior, and Basic Biology Lab., P. O. Box 14565, Gainesville, FL 32604)

Swarms of *Aedes taeniorhynchus* mosquitos produce sounds that can be detected from 10–50 m at 25–35 dB ($\approx 20 \mu\text{Pa}$) in a quiet environment. The loudness of the swarms is above the 21-dB acoustical background between frequencies of 0.6 and 3.4 kHz in an isolated salt marsh, but below the 40–60 dB background of a typical urban environment. Individual *Ae. taeniorhynchus* have wingbeat frequencies (\pm standard error) of 441 ± 21 Hz and 703 ± 17 at 24°C, detectable from 3 cm at 22–25 dB in an anechoic chamber. In the marsh, females beat their wings at 400–500 Hz and males at > 800 Hz, depending on their size and the temperature. Because of their low wingbeat sound intensities, the only individual *Ae. taeniorhynchus* mosquitoes that can be detected are those flying within 2–5 cm of the microphone. These frequency and loudness measurements suggest that it is technologically feasible to construct an acoustical device for remote surveillance of *Ae. taeniorhynchus* swarms or individuals attracted to a bait. Further studies to correlate detection of wingbeat sounds with numbers of mosquitoes captured at a bait are in progress.

2:00

3pAB5. Acoustic features of tonal "grunt" calls in baboons. Michael J. Owren, Christopher D. Linker, and Matthew P. Rowe (Dept. of Psychol., Univ. of Colorado, Campus Box 173, P. O. Box 173364, Denver, CO 80217-3364)

The acoustic features of "grunt" calls recorded from free-ranging baboons (*Papio ursinus*) in Botswana were investigated. Analysis parameters typical of those used to analyze human speech were employed

and these sounds were found to closely resemble speech vowels in several respects. They were brief (approximately 97 ms) and highly tonal with a stable fundamental frequency (approximately 122 Hz.) Like adult human males, baboons are estimated to have a vocal tract length of 17 cm (glottis to lips) and this concordance was reflected in the formant structure of the grunts. Three formants (range 2 to 7) were

typically found between 0 and 5 kHz, in a pattern resembling that of the vowel "schwa." Variation due to individual caller and behavioral context are discussed, as well as the implications of these findings for relationships between nonhuman primate vocal communication and human speech.

WEDNESDAY AFTERNOON, 6 OCTOBER 1993

SILVER ROOM, 1:00 TO 3:20 P.M.

Session 3pPA

Physical Acoustics: Observing Scattering with Time-Frequency Representations

Charles F. Gaumont, Chair

Acoustics Division, Physical Acoustics Branch, Naval Research Laboratory, Washington, DC 20375-5000

Chair's Introduction—1:00

Invited Papers

1:05

3pPA1. Time-frequency description of signals—A review. Leon Cohen (Dept. of Phys., Hunter College and Graduate Center of CUNY, New York, NY 10021)

The frequency content of many natural and man-made signals changes drastically with time and standard Fourier analysis does not fully describe such highly nonstationary signals. Among such signals are speech, sonar and radar, optical images, and biological and geophysical signals. In the past 10 years there have been dramatic strides made in our ability to understand and process such signals. The basic idea is to develop a method that describes the intensity of a signal jointly in time and frequency. This would give the frequency content at each instant of time and hence describe how the spectrum is changing in time. A review is presented of the ideas and methods that have been developed to describe a time-varying spectrum and their application is illustrated with concrete examples from a wide variety of fields.

1:35

3pPA2. Decomposition of scattered waves from an isolated elastic object with time-frequency representation. Nai-chyuan Yen (Phys. Acoust. Branch, Naval Res. Lab., Washington, DC 20375-5320)

In observing the scattering phenomena from an isolated elastic object, one would like to understand the composition of the scattered waves so the characteristic structure of the scatterer can be estimated. This paper reviews such an inverse scattering problem in terms of wave packet decomposition. Particularly, the focus is on a heuristic approach that has been developed from the study of acoustic scattering. The basic scheme of this approach is to examine the coherence property of the wave signature of the scattered waves and to compute its energy distribution in a joint time and frequency representation. Also shown is how to apply signal synthesis techniques to determine the dominant components, namely wave packets, expressed as the natural frame (basis), the building block for the formulation of the scattered wave. The magnitude and the phase of each wave packet are evaluated by algorithms that treat them as the independent nonorthogonal components from a set of linear equations. Examples of the response of the scattered waves from shell type elastic objects are used to illustrate the physical implication of this decomposition method in resolving the inverse scattering problem. It is concluded that the wave packet decomposition by means of the natural frame (basis) can be considered as a generalized Gabor and/or wavelet transform. Their relationships are discussed through the fundamentals of linear functional analysis.

Contributed Papers

2:05

3pPA3. Time-frequency spectrograms of impulse scattering by shells: Quantitative comparisons with ray theory for Lamb wave contributions. D. H. Hughes (815 Main St., Boonville, MO 65233) and P. L. Marston (Washington State Univ., Pullman, WA 99164-2814)

The Wigner distribution function (WDF) and quantities derived from it were used to investigate high-frequency scattering processes for a spherical shell in water [D. H. Hughes, Ph.D. thesis, Wash. State Univ. (1992)]. The investigation includes: (i) display of Gaussian smoothed WDF for ka as large as 500 with smoothing functions of

different time and ka resolution; (ii) measurements of the leaky Lamb-wave decay rates from the smoothed WDF for different ka regions showing good agreement with ray theory based on complex partial-wave index poles; and (iii) evaluation of a derived quantity, the local temporal variance of the WDF as a function of frequency, showing how the temporal spread changes near a resonance [D. H. Hughes and P. L. Marston, *J. Acoust. Soc. Am.* **94**, 499-505 (1993)]. The temporal variance at resonance was in general agreement with the ray theory predictions for the dominant scattering contribution. Structure in the smoothed WDF was also investigated for the prompt backward wave contribution that causes a prominent scattering enhancement at high frequencies. [Work supported by ONR.]

2:20

3pPA4. The prediction of echoes from underwater structures in the time-frequency domain. G. Gaunard (Naval Surface Warfare Ctr., White Oak Detachment, R14, Silver Spring, MD 20903-5640) and H. Strifors (Natl. Defense Res. Establishment, Sundbyberg, Sweden)

An earlier study [J. Acoust. Soc. Am. **93**, 2412 (A) (1993)] of echoes from submerged elastic shells in the combined time-frequency ($t-f$) domain is continued. Wigner and pseudo-Wigner distributions as well as time-windowed Fourier transforms are used [cf. L. Cohen, Proc. IEEE **77**, 941 (1989)]. These distributions extract from the echoes the time evolution of the target resonances that may be present in the frequency band of operation. Selecting suitable carrier frequencies for the incident pulses and appropriate spectral windows, it is possible to produce ($t-f$) displays linking resonance features to specific target characteristics. It becomes clear that this type of display is helpful in the processing of the echoes, and in the identification of the scatterer's material properties and geometrical shape. Broader/narrower time windows control the amount of detail in the resulting ($t-f$) plots, which can be adjusted at will. This is illustrated with examples exhibiting results in a colorized three-dimensional format as beautiful as practical and informative. These examples include the use of various realistic incident pulses, targets, and viewing windows that highlight the target-ID capability of the approach.

2:35

3pPA5. Time, frequency, and thickness analysis of monostatic scattering from an infinite steel cylindrical shell immersed in water. Timothy J. Yoder (SFA, Inc., 1401 McCormick Dr., Landover, MD 20785), Louis R. Dragonette, and Charles Gaumont (Naval Res. Lab., Washington, DC 20375)

The monostatic scattering from an infinite cylindrical steel shell is analyzed for frequencies up to $ka=10.0$. The shell thickness is varied from $h/a=0.001$ to $h/a=0.999$. The time, frequency, and thickness dependence of the scattering is graphically analyzed to determine the physical process(es) responsible for the scattering. This analysis reveals the following processes: (a) The Lamb wave circumnavigating the cylinder; (b) a 180-deg phase shift in the specular scattering as the shell becomes thick; (c) the coincidence frequency of the flexural waves with the radiation medium; (d) a drastic decrease in the " Q " of the modes supporting the flexural waves caused when the circumnavigating flexural waves begin to radiate efficiently; (e) the onset of the creeping waves as the shell becomes thick; (f) the transition of flexural/Lamb

wave behavior to Rayleigh wave behavior as the shell becomes a solid cylinder; (g) cancellation of the monopole and dipole modes for frequencies less than $ka=3$ occurring at thickness $h/a=0.05$; (h) the poles of the normal mode series solution coinciding with the highlights seen in the frequency domain.

2:50

3pPA6. Time-frequency representation of scattering from a target in a bounded medium. Angie Sarkissian, Charles F. Gaumont (Naval Res. Lab., Washington, DC 20375-5350), and Timothy J. Yoder (SFA, Inc., Landover, MD 20785)

Time-frequency representations applied to target scattering enable one to observe target characteristics that are of particular interest in inverse scattering problems. In this work such representations are extended to targets placed in a shallow water environment where target characteristics as well as shallow water characteristics appear in the time-frequency domain. The Gabor transformation is applied to scattering from an elastic shell placed in a bounded medium to display the combination of the waveguide effects (shallow water propagation) and the target effects (elastic wave propagation). The use of such representations for classification purposes is discussed.

3:05

3pPA7. Adaptive wavelet analysis of transient signals in structural acoustics. P. R. Stepanishen (Dept. of Ocean Eng., Univ. of Rhode Island, Kingston, RI 02881), Q. Huynh, and T. Greene (Naval Undersea Warfare Ctr., Newport, RI 02841)

Linear and quadratic time-frequency transforms are used to investigate and characterize transient signals from several wave propagation problems in structural acoustics. The transforms include: the short-time spectrum, Wigner type representations, and wavelet methods. Advantages and disadvantages of the various representations within the present context are presented. Particular emphasis is placed on using the wavelet packet method of Coifman to analyze and characterize complex transient signals from various structural-acoustic transmission paths with dispersive characteristics. Adaptive procedures are used to determine optimal time-frequency localization by using the best basis functions for the class of dispersive signals of interest. Various features and parameter extraction methods based on the local maxima of the wavelet transform at each scale are presented for several complex transient dispersive signals. [Work supported by ONR.]

WEDNESDAY AFTERNOON, 6 OCTOBER 1993

COLUMBINE ROOM, 1:25 TO 3:30 P.M.

Session 3pUW

Underwater Acoustics: Shallow Water Noise II

Kenneth E. Gilbert, Chair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Chair's Introduction—1:25

Invited Papers

1:30

3pUW1. Ambient noise in shallow seas: Propagation issues. Ira Dyer (Dept. of Ocean Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Whatever the source of ambient noise in a shallow sea, propagation to a distant receiver shapes in a major way its spectral character, its temporal properties, and its coherence or spatial distribution functions. Any one ambient noise mechanism is generally an incoherent superposition of a large number of similar elemental sources. Thus propagation modeling for ambient

noise must address the multiple source question, and as well the modeling can take advantage of simplifications often afforded by multiple path averaging. In this context, the main confounding elements of shallow sea propagation including, but not limited to, downward refraction, sloping bottoms, sediment interaction, rough bottom scattering, and water column fluctuations are reviewed. In such complicated propagation channels the noise is modified in important detail, although much of its character remains broadly recognizable.

1:55

3pUW2. Spatial structure of ambient noise in shallow water. Michael J. Buckingham (Marine Phys. Lab., Scripps Inst. of Oceanog., La Jolla, CA 92093-0213)

Apart from the nature of the sources, ambient noise in shallow water is distinguished from that in deep water through the proximity of the bottom. As with acoustic propagation, the bottom introduces a modal structure into the noise, evident through noise eigenrays with discrete grazing angles that are characteristic of the modes. There is also a continuous noise component, associated with overhead sources. Since it is largely controlled by the bottom, the modal structure is characterized by the geoacoustic properties, including the stratification, of the sediment. As the modal structure determines the directionality, or equivalently the spatial coherence in the vertical, it should be possible, through measurements with vertically aligned hydrophones, to establish the bottom properties through measurements of the noise field in the water column. For example, the sound speed of a fluid sediment close to the water-seafloor interface can be estimated from observations of the vertical noise coherence over the frequency range 500 Hz to 1 kHz (M. J. Buckingham and S. A. S. Jones, *J. Acoust. Soc. Am.* **81**, 938-946 (1987)). This inversion technique can be extended to the case of a shear supporting bottom and, at sufficiently low frequency, may yield information on the sub-bottom structure. [Research supported by ONR.]

2:20-2:30 Break

2:30-3:30

PANEL DISCUSSION:

Panel Members: I. Dyer, M. J. Buckingham

WEDNESDAY AFTERNOON, 6 OCTOBER 1993

**BOETTCHER CONCERT HALL, DENVER CENTER FOR THE PERFORMING ARTS,
3:30 P.M. TO 4:30 P.M.**

Performance/Lecture

Voices of People and Machines

Ingo Titze, Presenter

*Recording and Research Ctr., Denver Center for the Performing Arts, 1245 Champa Street, Denver, Colorado 80204 and
Wendell Johnson Speech and Hearing Ctr., University of Iowa, Iowa City, Iowa 52242*

WEDNESDAY AFTERNOON, 6 OCTOBER 1993

BOETTCHER CONCERT HALL, DENVER CENTER FOR THE PERFORMING ARTS, 4:30 P.M.

Plenary Session: Business Meeting and Awards Ceremony

Richard H. Lyon, Chair
President, Acoustical Society of America

Business Meeting

Presentation of Awards

Pioneers of Underwater Acoustics Medal to Homer P. Bucker

Silver Medal in Acoustical Oceanography to Clarence S. Clay

Silver Medal in Physical Acoustics and Engineering Acoustics to Steven L. Garrett

Session 4aAA

Architectural Acoustics and Engineering Acoustics: Teleconferencing Facilities

David L. Adams, Cochair

David L. Adams Associates, Inc., 1701 Boulder Street, Denver, Colorado 80211

James E. West, Cochair

Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, New Jersey 07974

Chair's Introduction—8:30

Invited Papers

8:35

4aAA1. Acoustic echo cancellation for stereophonic teleconferencing. M. Mohan Sondhi, Dennis R. Morgan, and Joseph L. Hall (Acoust. Res. Dept., AT&T Bell Labs., Murray Hill, NJ 07974-0636)

In long-distance telephony, echoes arise due to impedance mismatches at various points in the telephone circuit. Adaptive line echo cancelers have been used successfully for over a decade to combat this problem. Echoes also arise in teleconferencing, due to acoustic coupling between microphone and loudspeaker in each conference room. This problem is similar to the line echo problem; however, the echo paths are much longer and much more variable in this case. In this paper a further complication that arises if stereophonic transmission is used for teleconferencing is discussed: There is an inherent nonuniqueness in estimating the echo paths. It appears that the only way to resolve this nonuniqueness is by somehow decorrelating the signals in the two stereo channels. Several methods of decorrelation are discussed and how they affect adaptive echo canceller performance as well as stereophonic perception is shown.

9:00

4aAA2. New systems for intercommunications—Synergy of acoustics and telecommunications. Mark A. Holden (Jaffe Holden Scarbrough Acoust., Inc., 114A Washington St., Norwalk, CT 06854)

Proper room acoustics for audio teleconferencing and video teleconferencing facilities has been, at best, a matter of guesswork. In this paper, room acoustic treatments are varied in material and location within a typical conference room and the results studied using binaural analysis techniques. The relationship between room acoustics, speech intelligibility, speech interference, and mechanical noise levels in regard to teleconferencing systems will be explored in a quantitative manner, including the effects of "echo cancelling" technology.

9:25

4aAA3. A new approach for measurement telephone systems and teleconferencing applications. H. W. Gierlich (HEAD Acoust. GmbH, Kaiserstr. 100, 5120 Herzogenrath 3, Germany)

New telephone and teleconference systems show increasingly nonlinear time variant behavior. Measurement results of such systems are highly correlated to the measurement procedure used especially for conformance tests and measurements according to several standards. It is highly desirable to achieve comparable measurement results in different labs. These results should correlate well to the subjective impression the user has when telephoning with the measured device. To achieve the desired performance, a suitable test signal must have voice signal properties. On the other hand, measurements have to be carried out in a very short period of time in order to guarantee quasistationary behavior during the time of measurement. To achieve this, a "composite source signal" was developed. The signal consists of a voiced sound on 50 ms at the initial phase. This voiced sound is followed by a measurement signal, a pseudorandom noise sequence. To get a modulation typical for speech, the signal is followed by 100-ms pause. Variations of the signal are described for different measurement applications. Measurement results derived from different systems, including various signal processing techniques, are presented.

9:50

4aAA4. Architectural acoustics design parameters for teleconferencing facilities. Dennis A. Paoletti (Paoletti Associates, Inc., 40 Gold St., San Francisco, CA 94133)

Audio and video teleconferencing equipment and room facilities have emerged as one of the fastest growing areas of communication and business technology. The acoustic environment is an extremely important component in (determining) the success of the teleconferencing experience. Additionally, other environmental factors such as: architectural space planning, design aesthetics, color, thermal, lighting, operations, etc. must also be considered. This paper will present a summary of the significant acoustical parameters and key environmental factors that impact the design and functional use of audio and video teleconferencing facilities, whether teleconferencing equipment is portable or permanently built-in. (This paper acknowledges the work of

10:15

4aAA5. Audio for video teleconferencing. David A. Berkley (Acoust. Res. Dept., AT&T Bell Labs., 600 Mountain Ave., Murray Hill, NJ 07974-0636)

Video teleconferencing, especially that supported by inexpensive low-bit rate connections, is an exploding area. It is clear that there are severe trade-offs to be made as bit-rate is lowered. However, although everyone agrees that audio is a vital part of audio/video teleconferencing, little is understood about what trade-offs are involved in optimizing the audio and the relationship between the audio and video on a specific type of conference link. The relationship between audio and video transmission delay, camera tracking errors and lag, audio bandwidth and other related issues are discussed. Implementation of experiments, based on the HuMaNet [D. A. Berkley and J. L. Flanagan, AT&T Tech. J. 69, 87-97 (1990)] platform, is also described.

Contributed Papers

10:40

4aAA6. Acoustic aspects related to the performance of teleconference facilities. Bradley P. Basnett (Teleconferencing Dept. 4Y44, Bell-Northern Research Ltd., P.O. Box 3511 Station "C," Ottawa, ON K1Y 4H7, Canada)

Room acoustic design and electroacoustic system integration are paramount to the performance, and therefore effectiveness, of teleconferencing facilities. Some key elements that need to be considered during the design and implementation stages are (a) the effects of room acoustics on transmitted, received, and in-room speech communications; (b) the effect of transmission bandwidth on intelligibility and audio quality in the teleconference suite; (c) the naturalness of interaction over a telephone network; (d) the impact of acoustic and electroacoustic devices in a visual communications medium; (e) room ambient noise and potential noise sources. These elements are examined in light of the recent design, implementation, and operation of 12 teleconference (audio and video) suites. Details of design choices and acoustic measurements of these teleconference suites will be presented.

10:55

4aAA7. A binary method for measuring the electroacoustic characteristics of hands-free telephones. John R. Bareham (Consultant in Electroacoust., 20 Rogers Ave., Marlborough, MA 01752)

Hands-free telephones often are designed with performance features that make it difficult to obtain meaningful electroacoustic measurements using simple methods. Compressors, limiters, threshold detectors, noise guards, and other more complex speech-sensitive functions change the telephone's electroacoustic characteristics depending on the type and level of signal applied to it. To realistically account for such variables, electroacoustic characteristics of several hands-free telephones were measured using a binary method, in which two signals were applied to

the device simultaneously. The first signal was a bias, which temporarily caused the telephone to operate in a well-defined manner. Bias signals used in this work included noise, noise bursts, and speech fragments. The second signal was used for the actual measurement. It was applied at a low level relative to the bias, so the effect of the bias was not disturbed. The measurement signal used was a series of very fast sine sweeps, processed using an enhanced time delay spectrometry algorithm. This method was chosen to enable simulated free-field measurement of the telephone in an ordinary room. The measurement setup will be described, and measurement results will be shown and explained. Benefits and limitations of the method will be discussed.

11:10

4aAA8. A new artificial ear for telephone measurements. Winfried Krebber, Stefan Böhme [Inst. for Commun. Eng., Aachen Univ. of Technol. (RWTH), Melatener Str. 23, D-52056 Aachen, Germany], and Hans W. Gierlich (HEAD Acoustics GmbH, Herzogenrath, Germany)

Opinion tests show the importance of the acoustical leakage between the handset and the human ear for the subjective speech transmission quality of telephone sets. Artificial ears, which are now used for telephone measurements, do not allow measurements with variable leakage. Using an artificial ear formed as an individual human ear does not lead to reproducible results. Both problems are avoided by this new ear. It has a simplified shape, which is determined by only a few geometrical parameters. Thus the transmission characteristics can be described by a mathematical model. Nevertheless it approximates a human ear, referring to the dependence of the leakage and the transmission characteristics on the force applied to the handset. Measurements using the artificial ear are reproducible and show high conformity with measurements of handsets at human ears. Implemented in an artificial head, the ear can also be used to measure hands-free telephones. The directional characteristics of the artificial ear in a free sound field show all main structures of monaural outer ear transfer functions.

Session 4aAB

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Effects of Noise on Marine Mammals I

W. John Richardson, Chair

LGL Limited, 22 Fisher Street, P.O. Box 280, King City, Ontario L0G 1K0, Canada

Chair's Introduction—8:25

Invited Papers

8:30

4aAB1. Behavioral and hearing responses of pinnipeds to rocket launch noise and sonic boom. Brent S. Stewart (Hubbs-Sea World Res. Inst., 1700 South Shores Rd., San Diego, CA 92109)

Loud launch noise and sonic booms from some military space vehicle launches from Vandenberg Air Force Base impact pinnipeds on the mainland and at San Miguel Island, California, and may cause stampedes, temporary threshold shift or, less likely, permanent hearing damage. Maximum fast A-weighted (MFXA) sound levels approximately 4.8 miles downrange during the launch of three Titan IV rockets were 93.2, 92.7, and 93.0 dB; average sound exposure levels were between 98.9 and 101 dB. Harbor seals fled into the water in response but many returned to land within several hours. A sonic boom was recorded at San Miguel Island during one launch. Its peak flat sound-pressure level was 129.5 dB and maximum fast A-weighted sound level was 86.2 dB; virtually all of the energy was below 500 Hz. Pinniped behavioral responses were mild and brief. Predicted overpressures for focused sonic booms from the Titan IV rocket are substantially greater (≥ 150 dB). Noninvasive hearing tests using ABR and OAE techniques are being used to determine if pinnipeds suffer temporary threshold shifts or permanent hearing damage from exposure to those sonic booms.

8:45

4aAB2. Walrus response to offshore drilling operations. Jay Brueggeman (Ebasco Environmental, 10900 N.E. 8th St., Bellevue, WA 98004)

Walrus response to drilling operations in the Chukchi Sea was evaluated between 25 June and 19 October, 1989. Aerial and vessel observations of walrus were conducted at three prospects in conjunction with acoustic measurements of the operations. Walrus response was evaluated before, during, and after they passed the drillsite relative to various sound sources. Over 350 groups comprising approximately 4500 walrus were observed in the prospects. Walrus response was greatest during ice management, when the icebreaker crisscrossed the prospect. Animals moved deeper into the pack ice, where the noise level from the icebreaker was an estimated 15–25 dB above ambient (97 dB). Once ice management stopped or became more focused at the drillsite, walrus began to reoccupy formerly used areas. Under these circumstances, walrus displayed some behavioral responses that rapidly decreased beyond 0.46 km (0.25 nmi) from the icebreaker. Walrus showed little response to other drilling operations. These results show that walrus reacted to icebreaker activities, but responses varied according to the intensity of ice management. This variability offers opportunities to incorporate precautions to minimize disturbance to walrus during future drilling operations.

9:00

4aAB3. Experiments with an acoustic harassment system to limit seal movements. Bruce Mate (Hatfield Marine Sci. Ctr., Oregon State Univ., Newport, OR 97365)

To control the distribution of seals around salmon hatcheries, pond tests were conducted using swept frequencies between 2 and 20 kHz. This did not affect salmonid movements or reproduction. Aperiodic 12- or 17-kHz pulses of varying duration were effective at levels of 187 dB *re*: 1 μ Pa in reducing seal abundance near several hatcheries and pen aquaculture facilities. A few larger (possibly older) seals habituated or were less sensitive, and foraged with modified techniques. For sea lions, the same system produced a dramatic initial startle response but was otherwise totally ineffective. Many marine mammals react to moving sound sources even if loud stationary sources are tolerated. Early in this experimentation, swept frequencies were eliminated for simplicity. However, the illusion of motion as simulated by Doppler-like sweeps may have been lost in the process. Operant conditioning research suggests that an aversive stimulus is best maintained when used aperiodically. If several stimuli have a deterrent effect, this effect can be sustained longer by intermixing the stimuli to avoid or delay habituation. These principles may also be applicable to intentional harassment and industrial effects.

9:15

4aAB4. Long-range responses of belugas and narwhals to ice-breaking ships in the Northwest Passage. Kerwin J. Finley¹⁾ (LGL Ltd., Environmental Res. Associates, P.O. Box 280, King City, ON L0G 1K0, Canada) and Charles R. Greene (Greeneridge Sci., Inc., Santa Barbara, CA 93110)

Responses of belugas and narwhals to ice-breaking ships were studied during three spring seasons. The two species reacted differently. Belugas exhibited panic as ships approached at distances of 35–50 km; received noise levels ranged from 94–105 dB re: 1 μ Pa (20- to 1000-Hz band). Sound spectrographs revealed a tone at 105 Hz from an ore-carrier ship at 130 km. Presumed alarm calls indicated that belugas were aware of the ore carrier at 85 km (101 dB, 20- to 1000-Hz band level); they moved up to 80 km away from ice-breaking operations. In contrast, narwhals showed subtle “freeze” responses and dispersed slowly. Ship approaches caused narwhals to cease vocalizing temporarily whereas belugas emitted noisy alarm calls. Narwhals returned to disturbance areas faster than belugas and resumed normal activities when received noise levels from ice-breaking operations were as high as 120 dB. These responses occurred at unprecedented ranges, but no previous field studies had been conducted in areas with marine mammal populations unaccustomed to industrial noises. The disparity between field observations and theory, based on beluga audiograms, is examined. Results of Finley *et al.* [Can. Bull. Fish. Aqua. Sci. 224 (1990)] are augmented by additional acoustical analyses. [Work supported by Canadian Dept. of Indian Affairs & Northern Development.] ^{a)}Present address: 10232 Summerset Place, Sidney V8L 4X2, Canada.

9:30

4aAB5. A model for assessing the impact of vessel noise on odontocete communication. Susan E. Cosens (Canada Dept. of Fisheries and Oceans, 501 University Cres., Winnipeg, MN R3T 2N6, Canada)

Offshore industrial noise sources ensonify marine mammal habitat and may mask detection of social signals and important environmental sounds. A probabilistic model of sound detection was developed for assessing the impact of underwater icebreaker noise on signal detection by beluga whales and narwhals. The probability of detecting sample signals, in the absence of vessel noise, was compared to that of detecting the same signals in ship noise, using the MV ARCTIC, as a sample noise source. Analysis of signal detection probabilities showed that loud signals centered on the 5-kHz critical band were more severely masked by ship noise than were quiet 5-kHz signals or loud 2-kHz signals. Thus long-range calls seem to be more susceptible to masking by ship noise than are short-range calls. The model could also be applied to other critical bands of interest. Changes in the probability of detecting vessel noise, as estimated by the model, were also correlated with changes in beluga and narwhal behavior, observed in response to the vessel in operation.

9:45

4aAB6. Hearing abilities of killer whales (*Orcinus orca*). David E. Bain (Marine World Foundation, 2001 Marine World Pkwy., Vallejo, CA 94589), Birgit Kriete (Univ. of British Columbia, Vancouver, BC V6T 2A2, Canada), and Marilyn E. Dahlheim (Natl. Marine Mammal Lab., Seattle, WA 98115)

A study of hearing abilities of killer whales was conducted to help assess the impact of noise produced by vessels. The study had five objectives: (1) determine the range of frequencies that killer whales can hear; (2) determine the ability to hear these frequencies in the presence of bandlimited masking noise; (3) determine whether vessel noise affects the ability to hear pure tones; (4) determine whether noise from different directions has different masking properties; and (5) determine the ability to detect simulated killer whale sounds in the presence of vessel noise. Data were collected using conditioned responses of four captive killer whales. Conclusions reached include: killer whales are sensitive to a wide range of frequencies (0.5–105 kHz); noise reduces the ability to detect signals of similar frequencies; very loud, low frequency, noise reduces the ability to detect signals even at much higher frequencies; noise has the strongest effect when it comes from in front of the whale, and the least effect when it is from the side or behind; and the ability to detect broadband signals such as killer whale calls and clicks is not substantially affected by low levels of vessel noise.

10:00–10:15 Break

10:15

4aAB7. The effects of noise on dolphin echolocation. Whitlow W. L. Au and Paul E. Nachtigall (Hawaii Inst. of Marine Biol. and Naval Command, Control and Ocean Surveillance Ctr., RDT&E Div., P.O. Box 1106, Kailua, HI 96734)

The echolocation capabilities of dolphins can be seriously affected by natural ambient and man-made noise. Many echolocation experiments have been conducted at our facility to study the effects of noise on the echolocation behavior and capabilities of dolphins. When a beluga was transported from San Diego Bay, California to Kaneohe Bay, Hawaii, the whale adjusted its sonar emissions to compensate for the louder ambient noise environment in Kaneohe Bay. It emitted higher intensity signals which had peak frequencies that were approximately one octave higher than in San Diego. Experiments with echo-locating dolphins have shown that their target detection and discrimination capabilities can be severely degraded by the introduction of masking noise. In most situations, the dolphins compensated for the presence of masking noise by emitting more clicks per scan and by increasing their signal intensity. The response latency defined as the time between the cessation of the last click in a click train and the touching of a response paddle also increased as the masking noise level increased. However, the effects of noise emanating from a specific location can be minimized by a dolphin utilizing the directional property of its auditory system.

10:30

4aAB8. Interaural time discrimination in the bottlenose dolphin. P. W. B. Moore and Deborah A. Pawloski (The Naval Command, Control and Ocean Surveillance Ctr., RDT&E Div., San Diego, CA 92152-6267 and SAIC, Kailua, HI 96734)

Binaural hearing is the advantage of two ears over one. Localization studies with marine mammals have shown that acoustic localization underwater is mediated by the same binaural mechanisms used by other mammals in air. Dolphin echolocation abilities are enhanced by the use of two separate receivers. Au and Moore [J. Acoust. Soc. Am. 75, 255–262 (1984)] modeled the animal's hearing system as a two-element array using receiving beam pattern measurements. The present study sought to measure

the interaural phenomena of interaural time difference discrimination (ITD) and interaural intensity difference discrimination (IID). A dolphin was trained to accept contact hydrophones, attached by suction cups, to the outer surface of the lower jaw in the area of the pan-bone and to perform detection threshold experiments. Audiometric data at several frequencies indicate that the dolphin minimum threshold for pure tones may be 20 dB different when measured with jaw phones. The first behavioral measures of ITD and IID discrimination and for a cetacean have been obtained. ITD and IID for clicks trains of various frequencies, presented with interclick intervals simulating various echolocation target ranges were collected. This method of collecting audiometric data represents a new approach for assessing bearing capabilities in cetaceans.

10:45

4aAB9. High-frequency hearing loss in old (25+ years old) male dolphins. S. H. Ridgway and D. A. Carder (Biosci. Div., Naval Command, Control and Ocean Surveillance Ctr., RDT&E Div., Code 5107B, 49620 Beluga Rd., Rm. 200, San Diego, CA 92152)

Eight bottlenose dolphins (four male, four female) were studied in an acoustic response time task [Ridgway *et al.*, *J. Acoust. Soc. Am.* **89**, 1967–1968 (A) (1991)]. The animals were trained to station 1 m below the water surface and respond to a stimulus tone (St) delivered through a hydrophone located 1 m in front of the blowhole. St duration varied from 60 to 450 ms and frequencies were 20, 40, 50, 60, 70, 80, 100, and 120 kHz. St were 120 dB (*re*: 1 μ Pa peak to peak at 1 m) \pm 24 dB in 6-dB steps. With the dolphin at 1-m depth and 1 m from the St hydrophone, the trainer started a randomly variable 3- to 20-St block. The computer selected St from a file in random sequence and interval (1.1 to 2.1 s in 0.1-s steps) and offered St via a St generator. Animal responses (ARs are the whistle or pulse train) were received by another hydrophone, digitized, and stored. Each AR file with 20 to 200 St was edited on a CRT display of a 700 ms St window. No-AR trials, noisy trials, and wrong ARs were identified. Three males (ages 25, 29, and 35) exhibited hearing disability at the four higher frequencies. One female (age 33) showed some loss above 80 kHz, but two females (age 32 and 36) responded to all frequencies as did a male (age 9) and a female (age 13).

11:00

4aAB10. Bioacoustics of baleen whales: From infrasonics to complex songs. Christopher W. Clark (Bioacoust. Res. Program, Cornell Lab. of Ornithol., 159 Sapsucker Woods Rd., Ithaca, NY 14850) and William T. Ellison (Marine Acoustics, Inc.)

One reason for concern over potential effects of man-made sounds on marine mammals stems from the fact that most baleen whales produce intense, low-frequency sounds. Complete descriptions of vocal repertoires are still lacking for many of the 11 species, and our understanding of how whales use sounds is poor. The application of passive array techniques has provided unique and exciting insights into the biological significance of whale sounds. Acoustic studies on bowhead whales migrating off Point Barrow, Alaska and recent work on pelagic species (including blue, finback, and minke whales) clearly demonstrate that whales are vocally active and produce a variety of signals. Whales counter-call, adopt distinctive call types when traveling in acoustic herds, sing complex songs using two simultaneous, nonharmonically related voices, and produce extremely long, patterned sequences of infrasonic calls. These results support the conclusion that sounds serve a variety of important communicative functions. There is also reason to believe that some whales actively use surface reverberation as a navigational cue. Thus the concern over the potential impact of man-made noise on the whales' acoustic behaviors is legitimate and this issue requires continued attention. [Work supported by ARPA, ONR, and the Department of Wildlife Management, Barrow, AK.]

11:15

4aAB11. Reactions of bottlenose dolphins, *Tursiops truncatus*, and migrating gray whales, *Eschrichtius robustus*, to experimental playback of low-frequency man-made noise. Peter L. Tyack (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Playback experiments of low-frequency noise to two species of cetacean were linked to measurements of transmission loss in order to relate responses to received sound level. Coastal observation sites were selected to allow visual tracking of animals using theodolites. The *Tursiops* playbacks involved a population resident near Sarasota, FL, using a 800–900 Hz M-code stimulus with a source level of 170 dB *re*: 1 μ Pa at 1 m. The identification of individual dolphins of known sex and age with predictable movement patterns permitted repeat playbacks to monitor habituation. The *Eschrichtius* playbacks were performed as the whales migrated past Big Sur, CA; each whale was only exposed once during its migration. A single airgun (source level = 226 dB) and several continuous (source levels 151–163 dB) stimuli associated with offshore oil industry were used. Observation of > 3500 migrating whales showed significant responses. Whales slowed down and altered course at ranges of 1–3 km, some near 0 dB SNR. Half of the whales avoided exposure to continuous stimuli at levels > 117–123 dB, to airgun pulses at levels > 170 dB. Bottlenosed dolphins exposed to continuous M-code at levels near 120 dB showed no such avoidance response.

11:30

4aAB12. Responses of gray whales, *Eschrichtius robustus*, to noise. Marilyn E. Dahlheim (Natl. Marine Mammal Lab., NMFS/AFSC, 7600 Sand Point Way N.E., Seattle, WA 98115)

Responses of gray whales to increased levels of noise were documented during playback experiments. Nine sound parameters were compared between control and experimental conditions: call rates, call types, frequency range (Hz), emphasized frequencies (Hz), received levels of sound (dB *re*: 1 μ Pa), call duration (s), calls exhibiting frequency modulation, pulses per series, and signal repetition rates. Whale surface behavior (i.e., dive durations, movements, and abundance) was also investigated. Analyses yielded: a description of call types; a characterization of acoustical habitats; and a determination of relationships between whale calls and habitats; and a determination of relationships between whale calls and habitats. Gray whales employed multiple behavioral strategies (e.g., modification of calling to optimize signal transmission and reception, movement out of the study area, etc.) to circumvent increased levels of noise in their environment. Responses varied with sound source and may also differ with geographical range and/or general behavior.

Session 4aEA

Engineering Acoustics: Underwater Transducers and Arrays

James M. Powers, Chair

Naval Undersea Warfare Center, Code 2131, New London, Connecticut 06320

Contributed Papers

9:00

4aEA1. Standard reversible transducer (USRD type E37) for high frequencies to 2.0 MHz at hydrostatic pressures to 20.7 kPa (3000 psi). James A. Tims and Larry E. Ivey (Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

A new pressure-stable reversible standard transducer, USRD E37, has been developed at the Naval Research Laboratory, Underwater Sound Reference Detachment for the frequency range 100 kHz to 2.0 MHz at hydrostatic pressure to 20.7 MPa (3000 psi) with operating temperatures from 5 to 35 °C within this frequency range. This USRD E37 maintains a smooth response over the broad frequency spectrum previously covered by both the USRD E27 and E8 standard transducers, which were not stable with temperature and static pressure. The E37 has been calibrated to 6894 kPa (1000 psi) at 3 °C. Its unique sensor configuration design uses a lead titanate disc sensor supported by a butyl rubber ring. The faces of the sensor are covered with PRC PR-1538 polyurethane and are held in place in the frame assembly by thin wire. The receiver preamplifier assembly is specially designed for high-frequency operation, with a built-in bypass relay to allow signals to be transmitted. The cylindrically designed outer housing is made of polycarbonate and has been statically pressure tested to 31.026 MPa (4500 psi).

9:15

4aEA2. Variational principle modeling of class IV flextensional transducers. Harold C. Robinson (Naval Undersea Warfare Ctr., New London Detachment, New London, CT 06320)

The variational principle is an approximation method that allows one to obtain accurate estimates of certain quantities using relatively crude trial functions for the physical behavior. This principle is applied to transducer analysis by coupling a variational principle developed for the driving element (including piezoelectric effects) to one for the shell. The motion of the transducer is, in turn, coupled to a variational principle for the surface pressure in a fluid medium. The modeling of a class IV flextensional transducer will be presented in detail. The *in vacuo* resonance frequencies for the piezoelectric driving stack and elliptical shell, as well as the *in vacuo* mode shapes for the driver-shell combination will be compared with existing theoretical and finite-element analyses. Variationally determined surface pressures for a single transducer will be presented and compared to finite-element results. Admittance plots and beam patterns for the single element will be shown. [This work was sponsored by the Office of Naval Research.]

9:30

4aEA3. Broadband impulse transducer for measurement of backscattering by objects in water. Christopher S. Kwiatkowski, Gregory Kaduchak, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A broadband impulse pressure transducer was developed using a 15×15 cm sheet of PVDF in water. Both sides of the sheet were in

contact with the water and subsequently larger sheets were used. A planar transient wavefront was achieved by applying a step voltage across the sheet. Prior to the arrival of waves from the edge of the sheet, the radiated wave was nearly unipolar and impulsive for frequencies up to 1 MHz. The transducer is useful for measuring the backscattering or reflected impulse response of objects in water because the backscattered or reflected wave is nearly completely transmitted through the sheet. Thus a small broadband receiver can be placed behind the sheet transducer. The transmission was demonstrated by measuring the impulse reflected by a plate and the transducer was used to measure the short-term impulse response by an elastic shell. After a sufficiently long time, the received impulse response is contaminated by edge contributions. [Work supported by ONR.]

9:45

4aEA4. Application of multilayered structure for flow noise reduction. Sung H. Ko (Naval Undersea Warfare Ctr. Detachment, New London, CT 06320)

Turbulent boundary layer pressure fluctuations can be reduced by filtering of its wave-number response with either a finite hydrophone or a hydrophone array, or by filtering the wave-number response through a layer (layers) of elastomer. The elastomer layer is frequently called an outer decoupler and the multilayered structure is a generalized outer decoupler. In general practice, hydrophone arrays are embedded within the outer decoupler to reduce the turbulent boundary layer pressure fluctuations and then mounted to an elastic structure. The analysis of a multilayered structure is presented here to evaluate the transfer function, which is a measure of wave-number filtering through an outer decoupler. The front of the first layer is exposed to turbulent flow and the back of the last layer is in contact with static air. The results presented here are the numerically calculated transfer functions and noise reductions achieved by various layers of elastomer mounted on a steel plate.

10:00

4aEA5. A great depth class V flextensional transducer design. Michel Letiche (Thomson Sintra ASM-525, Route des Dolines, BP157, 06903 Sophia Antipolis Cedex, France)

Flextensional transducers consist of a piezoelectric or magnetostrictive driver and a mechanical shell. The shell, excited on its first bending mode, amplifies the acoustic radiation. In a first part, the coupling between the driver and the shell is analyzed. The relative resonance frequency of each element determines the resonance frequency of the transducer and its radiated acoustic power. Furthermore, in most cases, operating depth is a significant limitation in the design of this kind of source—classical flextensional transducers on the work market are typically depth limited at about 200 or 300 m. To avoid the effects of the hydrostatic pressure, mechanical filters have been developed. The aim of these devices is to make the driver and the shell independent under static conditions. An equivalent scheme model is used to understand and to design this filtering element. Theoretical results are presented and a great depth class V flextensional transducer has been developed

using this original technology. Its acoustic and static behaviors have been optimized with an axisymmetric B.E.M./F.E.M. numerical code. Finally, theoretical results are discussed in comparison with experimental ones obtained with the CERDSM facilities in Castillon Lake. Finally acoustic performances of this low-frequency, high power, and great depth transducer are shown. [Work supported by DRET.]

10:15

4aEA6. Three-dimensional modeling of low-frequency active source arrays. Donald L. Cox and Robert T. Chapman (Naval Undersea Warfare Ctr., Div. Newport, Detachment New London, Code 423, New London, CT 06320)

A three-dimensional analysis of two closely spaced class IV flexensional transducers is presented. The analysis was performed using a finite-element program, NOMAD, developed to solve extremely large three-dimensional problems. The program NOMAD (NO Matrix Decomposition) uses the conjugate gradient iterative solution technique with element by element (EBE) storage and preconditioning and enhanced Cray vector processing. In order to demonstrate the modeling capabilities of the code, no symmetry downsizing of the two transducer model was invoked. Stress and flux density information are provided for the transducer response. For the acoustic media, both pressure and intensity information is presented. [Work supported by the NUWC IR/IED program.]

10:30

4aEA7. Understanding coupling between flexural disk transducers from near field measurement. Jean-Marc Cortambert^{a)} and Steven R. Baker (Phys. Dept. Naval Postgraduate School, Monterey, CA 93943)

Both in civilian and military applications of underwater acoustics the need of high-power, directional low-frequency sources has stimulated the development of volumetric arrays of closely spaced transducers. Understanding the coupling between transducers in the array becomes fundamental to proper beamforming in the active sonar array. The paper presents the study of two flexural double bilaminar disk transducers by the method of finite elements using the ATILA code. The modification of the vibration behavior and the near-field radiation distortion of an element induced by an adjacent element are computed. The results are compared with the measurement of the displacement on the flexural disk using surface mounted strain gages and the pressure field near the transducers. [Work supported by the French Delegation Generale pour l'Armement and the Naval Postgraduate School.]
^{a)}French Naval Engineer. Permanent address: DCN Toulon/SDET/BP77, Toulon 83000, France.

10:45

4aEA8. Linearizing the output of nonlinear transducers by preprocessing their input signals. S. E. Forsythe (Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856)

A technique has been developed at USRD to drive transducers to amplitudes at which nonlinearities in the output normally become evident while maintaining the output signal as a faithful copy of the drive signal. This is done by preprocessing the desired signal using a transducer model that includes the transducer's nonlinearity. The model is stated as a system of coupled ordinary differential equations (ODEs) which define the dynamic variables of the system (positions, velocities, accelerations) as well as the parameters of the system (masses, stiff-

nesses, viscosities, etc.). Instead of integrating the ODEs using the known drive, as would be done to predict the system's output, the ODEs are explicitly solved "backwards" given the desired output. This solution typically requires the accurate numerical calculation of derivatives as well as the point-by-point solution of simultaneous nonlinear equations. Techniques for the efficient solution of the equations as well as solubility issues are discussed. [Work support by Space & Naval Warfare Systems Command.]

11:00

4aEA9. A 350-m depth capable, air-compensated baffled transponder for the deep water portable tracking system. R. K. Menoche, R. J. Reid (Naval Undersea Warfare Ctr., Range Development Div., Newport, RI 02841), and P. J. Stein (Scientific Solutions, Inc., Nashua, NH 03062)

This presentation describes the design and testing of a novel baffled central up-link subsurface link (SSL) for the deep water portable tracking system (PTS). The central SSL is deployed at depths up to 350 m and collects the acoustic information from the bottom moored tracking transponders. These transponders can be deployed to depths down to 6000 m. Degradation of the up-link SSL performance is generally due to support ship radiated noise and surface generated ambient noise. These sources of interference can be suppressed with a downward looking receiver. This receiver must have a wide beamwidth (140 deg) and its response must fall off rapidly outside the beam to reject ambient noise. The device built consists of a hydrophone and projector mounted on the face of a steel signal conditioning plate (SCP). The plate is designed to have low vibration sensitivity. Behind the plate, and extending beyond the SCP edges, is a fiberglass air chamber. To prevent crushing, it is fed by an air-compensation bag. Extension of the air-backing beyond the SCP is important to ambient noise rejection. This soft surface helps to reject noise which diffracts around edge of the battle. The air-chamber design was chosen over material solutions because it was uncertain if, at the deeper depths, the latter could provide the pressure release surface required.

11:15

4aEA10. Reciprocity assessment of single transducers. Li-Feng Ge (Anhui Bureau of Technical Supervision, Hefei, Anhui 230001, People's Republic of China)

Although most conventional transducers are usually regarded as reciprocal at nominal signal level, it is still necessary to ascertain that the presumed reciprocal transducer is indeed reciprocal or possesses sufficient reciprocity in reciprocity calibration practice. A conventional reciprocity check involves a system comprised of the two transducers and the gaseous, liquid, or solid medium and its boundaries. This paper tries to give an absolute method for reciprocity assessment without the need of any other auxiliary transducer. The principle is that if the imaginary part of the electrical impedance of a transducer (i.e., the capacitance for piezoelectric-type or the inductance for electrodynamic-type transducers) varies linearly with its mass load impedance, the transducer is reciprocal. So, the linear correlation coefficient could be considered as a reciprocity index. The theory is applied to evaluate the reciprocity of a piezoelectric and an electrodynamic vibration generator. The coefficients are determined as 0.9915 (at 10 kHz) and 0.9773 (at 1 kHz), respectively. With the data processed by a complex least-square-fitting code [L.-F. Ge, J. Acoust. Soc. Am. **91**, 2326 (1992)], the relevant coefficients are adjusted to 0.9987 and 0.9724. Thereupon, the transducers can be regarded as reciprocal at their respective working frequencies. The assessment has been verified by the accurate calibration results.

Session 4aMU

Musical Acoustics: Wind Instruments and Synthesis by Physical Modelling

Douglas H. Keefe, Chair

School of Music, DN-10, University of Washington, Seattle, Washington 98195

Chair's Introduction—8:55

Invited Papers

9:00

4aMU1. Two complex effective lengths for air columns. R. Dean Ayers (Dept. of Phys. and Astron., California State Univ., Long Beach, CA 90840)

In the context of a plane-wave approximation for acoustic waves in an air column, the pressure reflection coefficient R is related to the relative input impedance Z/Z_c by a simple bilinear transformation. At any location along the air column, the complex effective length to an open, downstream end of an equivalent cylinder is defined as $L_o \equiv (j/2k_0)\ln(-R)$, where k_0 is the undamped propagation number, and that to a closed end is defined as $L_c \equiv (j/2k_0)\ln(R)$. The real parts of these quantities are just the conventional effective lengths to the corresponding ends, which have been used in the study of musical wind instruments. The imaginary parts incorporate the effects of damping within the air column and at its downstream termination. The common theme here is a simple extrapolation of interfering, damped plane waves, with k_0 serving as an artificially large attenuation constant. L_c turns out to be more useful than L_o for analyzing the brass instruments [R. D. Ayers, *J. Acoust. Soc. Am. Suppl.* 1 **88**, S163 (1990)], and an earlier treatment of undamped horns made implicit use of a real L_c [R. W. Pyle, Jr., *J. Acoust. Soc. Am.* **57**, 1309–1317 (1975)]. This new treatment is more realistic, computationally simpler, and conceptually more straightforward.

9:30

4aMU2. Transient behavior of time-domain wind instrument models. Peter L. Hoekje (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50614-0150)

An important playing characteristic of a musical wind instrument is its transient response, which is well suited for calculation by time-domain models. Typically, the instrument air column is described by a time-domain response (either impulse response or reflection function), and a reed valve by a flow control equation and harmonic oscillator dynamics. Since important musical effects have been observed for changes in harmonicity of instrument resonances on the order of 10 cents (0.6%), calculations intended to show such effects need to specify the instrument response with appropriate resolution. In one calculational experiment, the air column resonance frequencies are adjusted in the frequency domain, then the corresponding time-domain response is calculated and the dynamical model run. The results show significant changes in transient response due to minor changes in air column response or in reed resonance. In addition to FFT methods, the transient behavior of spectrum partials may be tracked by applying the Hilbert transform to individually filtered harmonics and finding their instantaneous frequencies and amplitudes. Alternatively, a system of nearly harmonic oscillators with varying phases, frequencies, and amplitudes may be fit to the transient waveform. Each of these methods has unique benefits and difficulties.

10:00

4aMU3. Adding pulsed noise to a flute physical model. Chris Chafe (Ctr. for Comput. Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA 94305)

Pulsed noise has been detected in the residual of steady flute tones after elimination of purely periodic components. LMS adaptive linear periodic prediction was used to trace the waveform through its slight period-to-period fluctuations. The predicted signal was removed from the original leaving a breathy sounding residual to examine. Noise pulses in musical oscillators result from period synchronous gating of the driving means. Bowed string instruments exhibit noise pulses arising from alternating stick-slip motion, where noise is introduced only when the string is slipping. Distinct pulses are also exhibited by the saxophone in which the reed modulates air friction. Flute noise is more continuous than in string or reed tones. Short time Fourier transformation of the residual signal reveals that pulses are present, but spectrally weighted toward higher frequencies. A physical model of the flute incorporating a corresponding noise synthesis method is presented. Results of the simulation are compared for pulse quality and effect on frequency jitter.

10:30

4aMU4. The influence of clarinet and saxophone reed responses on sound production. Douglas H. Keefe (School of Music, DN-10, Univ. of Washington, Seattle, WA 98195) and Serge Waeffler (Université du Maine, 72017 Le Mans Cedex, France)

Performers of reed-driven woodwinds understand the important role that the reed plays in the production of aesthetically expressive musical sounds, yet the current status of musical acoustics theory and experiment concerning the reed is ambiguous.

Benade (1976) stressed the importance of the reed resonance in controlling the tuning and tone color of reed-driven woodwinds, and Thompson (1979) experimentally measured such tuning effects using a metal reed with a quality factor (Q) of about 10. However, the Q of a clarinet reed has been reported to be about 3. Time-domain simulations of a clarinet with a Q of 3 did not show the predicted entrainment of the playing frequency to a subharmonic of the reed frequency (Keefe, 1992). Measured responses of moist clarinet and saxophone reeds using mechanical impact excitation have been obtained in a configuration in which the experimenter's thumb applies the static force of the reed against the mouthpiece. This is intended to mimic the lower lip of the player under playing conditions. Results indicate Q 's in the range of 5–10. Time-domain simulations of sound production demonstrate that entrainment can occur in these higher- Q regimes.

Contributed Papers

11:00

4aMU5. Input response of cylindrical and conical bores with a single tone hole: Measurements and predictions. Teresa Wilson and Douglas Keefe (Systematic Musicology Program, School of Music, DN-10, Univ. of Washington, Seattle, WA 98195)

There exist few measurements on the acoustical response of tone holes in conical-bore air columns. For practical application to woodwind design, the accuracy of predictive models is said to be insufficient [McCann and Mathews, *J. Acoust. Soc. Am.* **91**, 2449 (A) (1992)]. Experiments performed test the transmission line model of air columns and tone holes. The input impedance was measured over the range 150–10 000 Hz for a single tone hole in a cylindrical bore and in a conical bore. The cylindrical bore had length 920 mm and radius 7.2 mm, and the conical bore had length 630 mm, entry radius 6.5 mm, and exit radius 12.5 mm. In woodwinds, the low-frequency resonances control the playing frequencies. Hence, the resonance frequencies under 1 kHz were predicted from a model of the air column and tone hole. The measured and predicted resonance frequencies for the cylindrical bore agreed to within 6 ± 6 cents and for the conical bore agreed to within 23 ± 18 cents. The errors in the conical-bore theory are largest at low frequencies. Work is underway to refine the theory of tone holes in conical bores.

11:15

4aMU6. Dynamic spectral envelope synthesis of trumpet tones. James W. Beauchamp (School of Music, Univ. of Illinois, 2136 Music Bldg., 1114 W. Nevada, Urbana, IL 61801) and Andrew Horner (Hong Kong Univ. of Science and Technology, Kowloon, Hong Kong)

It is well known that spectral envelopes of brass tones vary with performed dynamic [D. Luce and M. Clark, *J. Acoust. Soc. Am.* **42**, 1232–1243 (1967)]. In this study average spectral envelopes were computed as functions of normalized instantaneous rms amplitude using time-variant spectral data from a group of 15 trumpet swell tones with

fundamental frequencies extending over a 2-octave range. It was found that 10 spectral envelopes, defined by amplitudes in each of 24 standard critical bands spanning the frequency range 0 to 10 000 Hz, were sufficient to characterize the tone set. Time-variant spectra were synthesized using the original rms amplitude-versus-time envelopes as driving functions to dynamically select and interpolate between the appropriate spectral envelopes for each moment of time. The time-variant spectra were then converted into sound signals using conventional additive synthesis. Since the synthetic tones sound almost identical to the originals, it appears that dynamic spectral envelopes are an important defining characteristic of trumpet tone behavior.

11:30

4aMU7. An experimental investigation of the phase difference between lip vibration and acoustic pressure in brass instrument mouthpieces. S. Yoshikawa (Fifth Res. Ctr., Defense Agency, Japan) and G. R. Plitnik (Univ. of Tsukuba, Tsukuba, Japan)

Over a century ago, Helmholtz classified reed instruments as those which tend to blow closed as the blowing pressure increases (the mechanical reed instruments), and those which tend to open as blowing pressure increases (the lip reed instruments). All subsequent work has tacitly assumed that these models are correct, and although measurements on clarinet-like systems support the inward-blowing model, the outward-blowing hypothesis has been widely used as the basis for theoretical models of the brass instrument's sounding mechanism without experimental verification. As an alternate theory, the hypothesis that the vibrating lips of a brass player may act as an inward-blowing reed has been investigated. For this preliminary study, the phase difference between lip vibration and acoustic pressure was measured in several different lip-driven devices: a cylindrical tube without a mouthpiece, a French horn mouthpiece coupled to a nonresonant practice pipe, and a trumpet. By means of this experiment study, it is hoped that a definite resolution to the operating mechanism for the vibrating lips of brass players will be effected.

Session 4aNS

Noise: Progress Report on the Continuing Activity Based on the May 1993 Special Workshop on the ASA's Role in Noise and its Control

Robert M. Hoover, Chair
Hoover and Keith, Inc., 11381 Meadowglen, Suite 1, Houston, Texas 77082

Invited Paper

10:15

4aNS1. Progress report on the continuing activity based on the May 1993 special workshop on the ASA's role in noise and its control. Robert M. Hoover (Hoover and Keith, Inc., 11381 Meadowglen, Ste. 1, Houston, TX 77082)

A discussion meeting is being sponsored by the Technical Committee on Noise to present the progress that has been made to date on the recommended actions that were developed by the working group participants of the May 1993 Special Workshop on the ASA's role in noise and its control. The results of the May workshop will be summarized. Representatives of each working group will present a discussion of the progress on specific initiatives that have been begun. These will include any collaborative actions being taken with the cooperation of other Technical and ASA Standing Committees.

Session 4aPA

Physical Acoustics: Physical Acoustics for Materials Characterization I

Leonard J. Bond, Chair
Center for Acoustics, Mechanics, and Materials, University of Colorado, Boulder, Colorado 80309-0427

Chair's Introduction—7:55

Invited Papers

8:00

4aPA1. Acoustic studies of high- T_c oxide superconductors. H. Ledbetter (Natl. Inst. of Standards and Technology, 325 Broadway, Boulder, CO 80303)

Various acoustic studies of high- T_c oxide superconductors, mainly cuprates, will be reviewed. Transition temperatures T_c range from about 20 K for the electron-conductor, $(\text{Pr-Ce})_2\text{CuO}_\Delta$, to about 125 K for thallium cuprate. Properties considered include sound velocities, elastic constants, attenuation, Debye characteristic temperature Θ_D , and Grüneisen parameter. Variables considered include composition, temperature, pressure, and magnetic field. Especially the polycrystals are complicated by defects such as twins, voids, microcracks, nonhomogeneities, texture, and impurity phases. Even monocrystals are seldom defect-free. Most studies focused on the $\text{YBa}_2\text{Cu}_3\text{O}_x$ compound, where x varies between 6 and 7. The bismuth cuprates behave differently from most other compounds, perhaps because of incommensurate structural modulations. No convincing evidence exists that the new superconductors show velocity-temperature or attenuation-temperature curves similar to conventional BCS superconductors. This suggests a mechanism different from the simple s -electron spherical-fermi-surface weak-coupling BCS model. However, the new mechanism involves phonons. A relationship between T_c and Θ_D has been established for four systems: La-O, Y-O, Bi-O, Tl-O. Interrelationships among acoustic properties and other physical properties such as thermal expansivity and specific heat are emphasized. It is emphasized that some acoustic properties, especially the bulk modulus, can be estimated from simple models. Finally, topics are recommended for further study.

4aPA2. Point-source/point-receiver ultrasonics: A versatile materials characterization tool. Wolfgang Sachse and Richard Weaver (Theor. and Appl. Mech., Cornell Univ., Ithaca, NY 14853-1503)

The essential elements of a *point-source/point-receiver* ultrasonic system include a broad bandwidth, small-aperture source generating elastic waves with a broad angular spectrum and a receiver with similar characteristics to detect them. By moving the source or receiver and stacking the received waveforms, one generates a *scan image* that provides a view of the complete elastic wave field in a test specimen. Wave arrivals are related to the speeds of propagation of various wave modes in various directions and the signal amplitudes reflect the propagational characteristics of the material. An interpretation of the scan images requires an understanding of the propagation of transient elastic waves in a bounded structure. Recent developments on both the forward and inverse problems are summarized and the full solution of the forward problem for the response of a general axisymmetric anisotropic viscoelastic plate is reported. The computed results are compared to measured waveforms in viscoelastic and transversely isotropic materials. Also reviewed are the approaches for processing single wave forms and scan images to recover the elastic and viscoelastic properties of a test specimen. Extensions of this approach to materials of arbitrary anisotropy and layered microstructures are also considered. [Work supported by Cornell's Materials Science Center (NSF) and the Office of Naval Research (Physical Acoustics and Solid Mechanics Programs).]

9:00

4aPA3. Ultrasonic tomography for materials characterization. David A. Hutchins (Dept. of Eng., Univ. of Warwick, Coventry CV4 7AL, UK)

Ultrasonic tomography may be used as a method for imaging the internal structure of a wide range of solid materials. To do so requires tomographic reconstruction programs that can deal with irregular geometries, acoustic raybending and the possible presence of velocity anisotropy. Image reconstruction techniques are now available to deal with such problems, and have been applied to several situations of interest to materials characterization. The first of these involves the study of samples in bulk solid form, where a scanning system involving sliding contact transducers has been used to collect ultrasonic data and image materials that have been subject to thermal cracking. Additional studies have investigated the failure of fiber-reinforced polymer composites under applied load. In both cases, correlation with acoustic emission (AE) data has been performed. Other work has used immersion and noncontact ultrasonic transduction to image the structure of thin plate material using Lamb waves. In all cases, images will be presented and conclusions drawn on the type of information available from such images. [Work supported by SERC.]

9:30

4aPA4. Ultrasonic nondestructive evaluation of a layered medium. V. K. Kinra, C. Zhu, P. Jamminet, and V. Iyer (Ctr. for Mech. of Composites, Dept. of Aerosp. Eng., Texas A&M Univ., College Station, TX 77843-3141)

This paper is concerned with developing a technique for an ultrasonic determination of the acoustical properties (thickness, wave speed, density, attenuation) of the individual layer in a layered medium. The layered medium is surrounded by an identical host medium (water) of infinite extent on either side. Normally incident plane longitudinal waves are considered. Transfer functions have been calculated in time domain as well as in frequency domain. A satisfactory comparison between the theory and the experiment was observed for a three-layered medium (forward problem). An inverse problem is defined as a simultaneous determination of several acoustic parameters from the measured transfer functions. A systematic analysis of the sensitivity of the transfer functions to the acoustic parameters has been carried out. The following inverse problems have been solved for a three-layered medium: three thicknesses for a three-layered medium; and wave speed and thickness of any one of the layers (given those of the remaining two layers). Finally, relative merits of the time-domain and the frequency-domain approaches have been studied. [Work supported by Texas Advanced Technology Program.]

10:00-10:15 Break

Contributed Papers

10:15

4aPA5. Elastic guided waves and the Floquet concept in periodically layered plates. D. E. Chimenti (Dept. of Aeroeng. and Eng. Mech., Iowa State Univ., Ames, IA 50011), P. J. Shull (Johns Hopkins Univ., Baltimore, MD 21218), and S. K. Datta (Univ. of Colorado, Boulder, CO 80303)

This paper describes the results of experiments and calculations on the interaction of ultrasound with aluminum/aramid-epoxy composites (ARALL™), a planar laminate of alternating constituent layers. The plates have been investigated experimentally using fluid-coupled leaky wave techniques, and theoretically using both a partial wave analysis and also a stiffness-based numerical approach. Frequency zones of wave transmission and reflection, similar to those predicted for periodic media have been observed and modeled in the bilayered plates. The onset

of this behavior has been simulated numerically by incrementally varying lamina properties starting from a homogeneous average value. It is demonstrated that the transmission zones arise from a consolidation or clustering of the uniform-plate minima into frequency zones or bands, similar to Floquet wave behavior. Moreover, these bands, and the associated dispersion curves, are approximately constant in frequency, entirely unlike the behavior expected for a homogeneous plate. It is suggested that the large phase shifts induced in a layered medium composed of elastically contrasting media are responsible for this observation. This mechanism also leads to a correspondence between the number of reflection minima and layer unit cells. The effect of structural symmetry on the reflection function has also been studied. The theoretical predictions compare very well, in nearly every case, to the experimental results.

4aPA6. Measurement of elastic properties of transversely isotropic plates using Rayleigh-Lamb waves. Wayne P. Rogers (Dept. of Mech. Eng., Univ. of Colorado, Boulder, CO 80309-0427)

This paper presents a novel method to experimentally determine the elastic constants of a transversely isotropic plate using measurements of phase velocity of Rayleigh-Lamb waves. The forward problem, i.e., to calculate the phase velocity versus frequency curves given known elastic constants, has been solved by numerical solution to the appropriate dispersion equations. The problem attacked here is the inverse problem, i.e., to determine the elastic constants given experimental measurements of phase velocity as a function of frequency. These elastic constants may be found by simultaneous solution of five distinct dispersion equations corresponding to particular measurements of phase velocity and frequency. This approach yields accurate solutions provided that appropriate Lamb waves modes, frequencies, and plate directions are chosen. The experimental method employs a pair of variable-angle-beam-contact transducers used in a pitch-catch mode with a harmonic wave phase comparison technique to measure phase velocity over the frequency range of 30 kHz-1 MHz. Results are presented for several isotropic plates and for a unidirectional-continuous-glass-fiber-reinforced epoxy plate. The elastic constants calculated using this inverse procedure are then used in the forward problem in order to compare experimental results to theoretical predictions of the dispersion curves for the first few modes. [Work supported by NASA-Langley.]

10:45

4aPA7. High-resolution velocity measurements in the superconducting state of the heavy fermion compound UPt_3 . S.-W. Lin, H. Zhang, M. Levy, Bimal K. Sarma (Phys. Dept., Univ. of Wisconsin, P.O. Box 413, Milwaukee, WI 53201), C. Jin, and D. M. Lee (Cornell Univ., Ithaca, NY 14853)

Ultrasonic measurements have been used very successfully in investigating multi-phase superconductivity in the heavy fermion compound UPt_3 . Further probing of the superconducting properties in UPt_3 demands even higher resolution in both ultrasonic velocity and attenuation measurements. In order to achieve this, a new approach was used in measuring ultrasonic velocity and attenuation, in which a digital dynamic control technique was utilized in the conventional pulse-echo method. With this approach, longitudinal ultrasonic velocity and attenuation of frequencies up to 450 MHz were measured on a single crystal UPt_3 sample in the superconducting state. The overall resolution in the velocity measurement is in the order of 0.1 ppm. This enables us to determine a complex superconducting phase diagram for UPt_3 with unprecedented accuracy. Superconducting phase boundary lines were traced even within the transition width. The critical property of converging phase transition lines is thus determined unambiguously. This in turn provides crucial information for the understanding of unconventional superconductivity in UPt_3 . [Work at UWM supported by ONR and at Cornell by NSF.]

11:00

4aPA8. Elastic constants and damping using ultrasonic resonance spectroscopy: Application to monocrystal and polycrystal copper. H. Ledbetter, C. M. Fortunko (Natl. Inst. of Standards and Technology, 325 Broadway, Boulder, CO 80303), and P. Heyliger (Colorado State Univ., Fort Collins, CO 80523)

Several authors used ultrasonic resonance spectroscopy to determine a material's elastic-stiffness constants, the Voigt C_{ij} . Complementary to these are the out-of-phase C_{ij}^* , which give the complete elastic stiffness $\bar{C}_{ij} = C_{ij} + iC_{ij}^*$. Here, C_{ij}^* measurements for both monocrystal and polycrystal copper are reported. For the monocrystal case, the C_{ij}^* vary strongly with the vibrational mode. And, when interpreted against a vibrating-string dislocation model, the C_{ij}^* present several surprises. For the polycrystal, the C_{ij}^* are much lower and nearly independent of vibration mode.

4aPA9. Radiative transfer and multiple scattering of diffuse ultrasound in polycrystalline media. Joseph A. Turner and Richard L. Weaver (Dept. of Theor. and Appl. Mech., 104 S. Wright St., Univ. of Illinois, Urbana, IL 61801)

A model for the multiply scattered incoherent field in a continuous polycrystalline elastic medium is presented. Unlike a previous development based upon energy and flux conservation considerations [J. A. Turner and R. L. Weaver, J. Acoust. Soc. Am. 93, 2312 (A) (1993)] for a medium containing discrete random scatterers, the present model has been developed from the wave equation and first principles. Appropriate ensemble averaging of the wave equation leads to Dyson and Bethe-Salpeter equations that govern the mean Green's function and the covariance of the Green's function, respectively. These equations are expanded for weak heterogeneity and equations of radiative transfer are obtained. The result is valid for attenuations that are small compared to wave number: $\alpha/k \ll 1$. Polarization effects are included, as before, through five elastodynamic Stokes parameters, one longitudinal and four shear. The theory is applied to a statistically homogeneous cubic polycrystalline half-space immersed in a fluid and illuminated by a plane wave. Results on the angular and temporal dependence of backscattered intensity are presented and compared with the predictions of a single-scattering theory. It is anticipated that this approach may be applicable to microstructural characterization through the study of the time, space, ultrasonic frequency, and angular dependence of multiply scattered ultrasound in elastic media. [Work supported by NSF.]

11:30

4aPA10. Guided waves in a fluid-solid bilayered plate. C. L. Yapura and V. K. Kinra (Ctr. for Mech. of Composites, Dept. of Aerosp. Eng., Texas A&M Univ., College Station, TX 77843-3141)

Consider a fluid-solid bilayer of infinite extent with the x_3 axis perpendicular to the bilayer and the material axes coinciding with the x_1 , x_2 , and x_3 axes. The fluid is assumed to be elastic (and isotropic) while the solid is assumed to be elastic and orthotropic (i.e., with nine independent elastic constants). Consider a plane wave propagating in the x_1 direction with the particle motion confined to the x_1 - x_3 plane and plane strain conditions in the x_2 direction. There are six boundary conditions: zero normal and shear stresses at the traction-free boundaries, zero shear stress and continuity of normal displacements, and normal stresses at the interface. The dispersion equation is obtained by setting the determinant of the resulting six-by-six matrix to zero. Numerical results in the form of phase velocity, group velocity, and mode shapes are presented for the case of a water/graphite-epoxy bilayer. Several interesting features are observed and discussed. The physics underlying these phenomena has been explored by studying the mode shapes at (and in the vicinity of) the frequencies of interest. [Work supported by Texas Advanced Technology Program.]

11:45

4aPA11. Wavelet representation of the three-dimensional anisotropic elastodynamic Green's function. Vinod K. Tewary and Christopher M. Fortunko (Mater. Reliability Div., NIST, 325 Broadway, Boulder, CO 80303)

A detailed knowledge of waveforms is required for nondestructive ultrasonic characterization of anisotropic materials. The waveforms can be calculated in terms of the elastodynamic Green's function of solids. The traditional Fourier/Laplace transform methods are computationally inefficient for three-dimensional anisotropic solids since they require four-dimensional numerical integration in the wave vector and the frequency space. A representation of the Green's function has been developed in terms of highly localized Huygens-type wavelets in which the Green's function is expressed in the space of slowness vectors rather than that of wave vectors and frequency. The wavelet representation requires only a one-dimensional numerical integration of simple functions and thus saves computational (CPU) time by a factor of about

1000 compared to the Fourier transform method. A solution of the tensorial elastodynamic Cauchy problem and calculation of the retarded Green's function will be described in terms of these wavelets. Results will be presented for pulse propagation in anisotropic solids.

12:00

4aPA12. Waveform-based ultrasonics in anisotropic hemispheres. Eric S. Boltz, Vinod K. Tewary, and Christopher M. Fortunko (Natl. Inst. of Standards and Technol., Mater. Reliability Div., Boulder, CO 80303)

Conventional ultrasonic materials characterization methods rely heavily upon measurements of acoustic wave velocities and amplitudes which are often ambiguous. Waveform-based ultrasonics, however, seeks to extract valuable information from the actual ultrasonic wave-

forms. This computational portion of the work seeks to mathematically represent ultrasonic waves propagating in a bounded, three-dimensional, anisotropic hemisphere. An integral representation for the exact solution of the Christoffel equation for wave propagation in anisotropic hemispheres is developed using a wavelet transform developed at NIST. This representation allows the computation of material displacements as a function of both position and time for a given source. This wavelet is a function of the slowness vector, rather than the wave vector, and is better suited to anisotropic solids where the direction of energy flow is parallel to the slowness vector. The source case considered is pencil lead break in the center of the flat side of an anisotropic hemisphere. Surface displacements are computed for various materials. Where possible, comparison with previous work and with experimental data are presented.

THURSDAY MORNING, 7 OCTOBER 1993

DENVER ROOM, 7:55 A.M. TO 12:00 NOON

Session 4aSA

Structural Acoustics and Vibration: Acoustic Tailoring of Materials and Structures

Courtney B. Burroughs, Chair

Engineering Science and Mechanics Department, Pennsylvania State University, State College, Pennsylvania 16804

Chair's Introduction—7:55

Invited Papers

8:00

4aSA1. Results of acoustic optimization of a point-driven, thin-shelled cube using the boundary element method. Daniel H. Kruger, J. Adin Mann, III, Amy Spalding, and Keith Holman (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50010)

The sound power radiated by a thin-shelled cube is calculated from simulated acoustic velocities (equal to the normal shell velocity) using the boundary element method (BEM) [K. A. Cunefare and G. H. Koopmann, *J. Vib. Acoust.* **113**, 387–394 (1991)]. Optimization techniques are then used to predict velocity distributions that result in the minimum output sound power. In future work, recommendations for the addition of passive noise control materials will be defined on the basis of achieving the calculated optimal velocity distribution that will result in the minimum radiated sound power. These criteria are tested on a point-driven, thin-shelled cube in an anechoic chamber. The normal shell velocity is measured using a laser velocimeter. Again, sound power is calculated and the optimal velocity distribution is determined using the BEM. Accuracy of power predictions, based on experimental power measurements and computational examples, as a function of frequency and vibration pattern are studied.

8:25

4aSA2. Some structural acoustic features of multilayer composite plates. M. Kim and Y. F. Hwang (Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

This paper discussed the propagation of free waves and the vibroacoustic response of a multilayer composite plate subjected to a point-force excitation. The dispersion characteristics for various modes of wave propagation as a function of geometrical configuration, elastic properties, and material loss factor are discussed. Acoustic radiation into water from a composite plate subjected to a point force acting on its dry (air) side has been evaluated. Numerical examples focus on a three-layer (elastic/viscoelastic/elastic) sandwich plate with a variety of different elastic/viscoelastic material properties (elastic constants, damping) and thicknesses. The advantages and disadvantages of a sandwich plate as compared to a homogeneous plate of equal flexural rigidity are also discussed. [Work supported by ONR.]

8:50

4aSA3. An efficient method for designing quiet composite structures via material tailoring. Jung Bae Oh and Gary H. Koopmann (Ctr. for Acoust. and Vib., Penn State Univ., University Park, PA 16802)

An efficient design method is successfully developed for achieving structures that radiate minimum acoustic power based on a material tailoring. First, the far-field acoustic power of a planar radiator is written as a quadratic expression in terms of the Rayleigh–Ritz formulation. Second, a complete acoustic power design sensitivity is developed analytically that is needed to find a search direction for the improved design. This analytical sensitivity significantly reduces computation time as compared to the

finite difference scheme and provides an exact design sensitivity information. The formulations for acoustic power and design sensitivity are coupled together within the numerical optimization code, CONMIN, to minimize acoustic power radiated from a planar surface subject to various excitation conditions via local thickness tailoring. The design of quiet structures for general conditions—harmonic point excitation, uniformly distributed harmonic acoustic loading, and random acoustic loading—in the given frequency band is achieved successfully. Results on various test cases show that the hybrid composite structure is the most promising material for minimizing radiated acoustic power. The optimum hybrid composite plate shows that the radiated acoustic power can be reduced up to 10 dB on various test cases.

9:15

4aSA4. Optimization of acoustical treatments with respect to dynamic compliance and static deformation under load. Edward M. Kerwin, Jr., Nathan C. Martin, Jeffrey A. Zimmer, and Jeffrey A. Doughty (Bolt Beranek and Newman, Inc., 70 Fawcett St., Cambridge, MA 02138)

This paper presents a method of organizing information on two important performance measures of an acoustical treatment or element whose dynamic compliance and total displacement under static loading are of concern. The method is simple but quite general, and allows one to select the optimum treatment from a set of candidates, or to modify and optimize a given candidate. The two normalized performance metrics are the unit-volume dynamic compliance q_v and a unit thickness-deformation parameter $V = \epsilon/q$, where ϵ is the static thickness strain under load. The optimization process utilizes these unit-volume parameters under the assumption that the required total dynamic compliance q_{tot} will be specified, although its numerical value need not be known ahead of time. In this case it follows that V is always proportional to the total thickness and total deformation under load. In addition the $V-q$ presentation has proven helpful in developing physical insight into the behavior of various compliant treatments. [Work supported by Office of Naval Research.]

9:40

4aSA5. Material tailoring of structures to achieve a minimum radiation efficiency. Koorosh Naghshineh (Acoust. and Radar Technol. Lab., SRI International, 333 Ravenswood Ave., Menlo Park, CA 94025), Gary H. Koopmann, and Ashok D. Belegundu (Penn State Univ., University Park, PA 16802)

A strategy is developed for designing structures that radiate sound inefficiently in light fluids. The problem is broken into two steps. First, given a frequency and overall geometry of the structure, a surface velocity distribution is found that produces a minimum radiation efficiency. This particular velocity distribution is referred to as the "weak radiator" velocity profile. A finite element adaptation of the integral wave equation is combined with the Lagrange multiplier theorem to obtain this surface velocity distribution. Second, a distribution of Young's modulus and density distribution is found for the structure such that it exhibits the weak radiator velocity profile as one of its mode shapes. Extensive use of structural finite element modeling as well as linear programming techniques is made to find this distribution. The result is a weak radiator structure. When compared to a structure with uniform material properties, the weak radiator structural response is found to exhibit lower wave-number content in the supersonic region. The effect of modal overlap on the performance of the weak radiator structures is found to be negligible. The example of a simple beam radiating in a rigid baffle is used for the purpose of illustration.

10:05–10:20 Break

10:20

4aSA6. Composite cylinder cross section tailoring for radiated noise reduction. Colin P. Ratcliffe (Dept. of Mech. Eng., USNA, Annapolis, MD 21402-5008), Roger M. Crane, and Armando L. Santiago (NSWC, Annapolis, MD 21402-5067)

This paper presents the results of an ongoing investigation into the reduction of transmitted noise for underwater cylindrical structures. A reduction in the radiated self-noise of cylindrical structures, compared with the conventionally utilized aluminum structures, was undertaken, investigating the use of composite materials with varying cross sections to meet the specified requirements. The cross sections investigated included several variations of a state-of-the-art triple skin construction, along with monolithic, constrained layer and a semi-debonded cross section. Cylinders have been fabricated and tested, comparing the dynamic vibration properties of the forced vibrations in air of the composite scaled cylinders to the baseline aluminum. The modal analysis determined mode shapes, natural frequencies, and damping loss factors. These were then tested in reverberant conditions under water to assess their acoustical performance. The in air and underwater tests were compared to identify similarities and differences between force excitation and acoustic excitation. The comparison showed where in air tests can be used as an aid to estimating underwater performance. The results also highlighted where in air tests are inadequate for predicting underwater behavior. The effect of the cross-sectional design for the tailoring of the acoustic signature will be discussed.

Contributed Papers

10:45

4aSA7. Acoustic reflections from acoustically tailored nonhomogeneous viscoelastic coatings. Courtney B. Burroughs and Kimberly L. Beach (Engineering Sci. and Mech. Dept., Penn State Univ., State College, PA 16804)

An analytic model for acoustic reflections from nonhomogeneous viscoelastic coatings is developed. The model includes variations in the viscoelastic properties of the coating in directions parallel to the surface

of the coating. By converting acoustic pressure wave incident on the coating to high wave-number pressure and shear waves in the coating, the nonhomogeneities in the coating convert potentially reflected waves into nonreflecting waves, thereby trapping the energy in the coating where the energy is dissipated by the internal damping in the coating. Examples are presented to show the sensitivity of acoustic reflections from coatings to the nonhomogeneous properties and damping of the coating material.

4aSA8. Structure-property relationships in polymer interpenetrating networks for use as broad band damping materials. Christopher S. Coughlin, Mary Q. Samuels, and Rodger N. Capps (Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

In the search for broadband damping materials, it is desirable to have polymers with a broad and high loss region in shear or extension, covering the entire temperature and frequency range of interest. Interpenetrating networks, IPN's, are materials composed of two or more crosslinked polymers intimately and irrevocably intertwined. The resulting distribution of microenvironments can result in a material with a high mechanical loss broadened over that of either polymer component alone. Several series of polyurethane/epoxy IPN's have been prepared for possible use as broadband damping materials. All IPN's showed apparent true IPN behavior with no or very small scale phase separation. Dynamic mechanical analysis revealed that the temperature of the loss peak may be varied over a wide temperature range with formulation. Further work has focused on incorporating a more flexible epoxy component and/or a plasticizer, and comparison of viscoelastic behavior to observed damping in constrained layer structures. [Work supported by ONR.]

11:15

4aSA9. Exploratory materials development for towed array sensor systems. Rodger N. Capps (U.S. Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

Extended towed array sensor systems are used for geophysical sensing and exploration in the petroleum industry, and for detection of submersible objects in naval applications. Many of the polymeric materials considered for use in towed array hoses either did not possess the necessary aging and weathering resistance; provided satisfactory self-noise performance at higher temperatures but were unacceptable in cold water; or experienced changes in self-noise through fill-fluid permeation and chemical reaction. There has been a need to develop and evaluate new classes of elastomers for use as hose wall materials. Array self-noise performance can also be improved through proper selection of polymers for use in vibration isolation modules for low-frequency towline excitation. Thermoplastic elastomers offer significant advantages over other types of elastomers in cost and ease of processing. Results are presented for both conventional elastomers and thermoplastics to show how these materials can be selected for the proper combination of Young's modulus and internal loss, tensile properties, ozone and weather resistance, resistance to swelling and permeation by seawater and fill fluids, adhesion to reinforcing members, for fabrication and evaluation as acoustic modules. Use of thermoplastics for vibration isolation modules is also discussed. [Work supported by NAVSEA PM 0425.]

4aSA10. New fiber-reinforced polyurethane composites as a soft sonar window. C. M. Thompson, R. Y. Ting (Naval Res. Lab.—USRD, P. O. Box 568337, Orlando, FL 32856-8337), and L. M. Leimer (Texas Res. Int., Austin, TX 78733)

A new polyurethane formulation has been developed for use as the acoustical window for a soft sonar dome. The polyurethane is based on a 3000 molecular weight difunctional polybutyleneoxyglycol (PBOG) softblock. In addition, a moderately high trifunctional PBOG was added to increase the cross-link density for improving the polymer toughness. "Spectra," an extended chain polyethylene fiber, was selected as the reinforcement fiber because it has excellent physical strength and is low in density. The lower density assists in matching the specific acoustical impedance of the composite to that of sea water. Fiber loadings up to 50% in thickness were used. Samples of this new composite were fabricated and their physical and acoustical properties were evaluated. Insertion loss of the test panel was measured in the USRD ATFI facility. The material was found to be acoustically transparent at the operating frequency range, and physically resilient to high impact loadings.

11:45

4aSA11. Development of a new rigid composite for high-frequency sonar window applications. C. M. Thompson, R. Y. Ting (Naval Res. Lab.—USRD, P. O. Box 568337, Orlando, FL 32856-8337), L. L. Jenkins, and C. C. Chiou (Texas Res. Intl., Austin, TX 78733-6201)

Successful operation of high-frequency sonar arrays requires an acoustic window that not only has a minimal interference with the acoustical signal but also is sufficiently rigid to protect the electronics from arctic ice penetration. Current designs of the window materials call for composite materials of epoxy containing either glass or Kevlar reinforcements. Those composites have unacceptably high insertion losses and beam pattern distortion. A new window material has therefore been developed based on a bromine-modified epoxy resin. The sound speed and the density of the composite were controlled by the bromine content and by using a soft microballon. The "Spectra" reinforcement fibers were used for meeting the strength requirement. Finite-element calculations shows that a three-layer structure of 1:3:2 Spectra/microballon/Spectra composite design should optimize the impact strength of the material while maintaining the acoustical performance. Samples of this new composite were fabricated and their mechanical properties evaluated, including the Izod impact strength. Insertion loss of test panels was measured in the USRD ATFI facility. The results of these tests will be presented for discussion. [Work sponsored by NAVSEA.]

Session 4aSP

Speech Communication: Current Speech Synthesis Technology

Joseph P. Olive, Chair

AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, New Jersey 07974

Chair's Introduction—9:00

Invited Papers

9:05

4aSP1. Text normalization and ambiguity resolution in speech synthesis. David Yarowsky (Dept. of Comput. and Inform. Sci., Univ. of Pennsylvania, Philadelphia, PA 19104)

Raw text contains many ambiguities that must be resolved in speech synthesis. Numbers and abbreviations often have different pronunciations in different contexts, as in the phrase "1240 people at 1240 St. Albans St." The token "IV" is pronounced differently in "Henry IV," "Section IV," and "IV drug." Proper names, acronyms, and words with the same part of speech may also have context-sensitive pronunciations. Previous speech synthesizers have used heuristics or simple defaults to handle these cases. In recent work [Sproat *et al.*, in International Conference on Spoken Language Processing (1992)], statistical decision procedures have been applied to this text normalization process. The talk will describe further developments in this work, using Bayesian discriminators and decision lists based on nearby words, word classes, and type of text as evidence for the selection between pronunciation variants. Examples will include implementations of these algorithms in the AT&T Bell Laboratories TTS synthesizer.

9:25

4aSP2. Text analysis for speech synthesis. Kenneth Ward Church (AT&T Bell Laboratories, 600 Mountain Ave., Murray Hill, NJ 07974)

Text analysis is a catch-all term for a range of tasks such as tokenizing the input text into words and sentences, assigning parts of speech identifying clitics, parsing phrasal verbs, identifying and expanding abbreviations, dates, fractions, and amounts of money, and so on. Text analysis is important for speech synthesis for two reasons. First, word pronunciation sometimes depends on usage: I can be a pronoun or a Roman numeral, wind can rhyme with "bind" or "binned," Dr. can be "doctor" or "drive," 2/3 can be "two thirds" or "February third," or "two slash three." A second, equally important reason for text analysis is that its results will be used to modulate the pitch, timing, and amplitude of the speech so as to present the text's message clearly. For example, the function word "that" should be cliticized (reduced) in certain usages (e.g., as a subordinating conjunction: "It is a shame that [schwa] he is leaving") but not in other usages (e.g., "Did you see THAT [no schwa]?"). If a synthesizer could reliably make such distinctions, it might sound a little more like it knows what it is talking about (and that it cares).

9:45

4aSP3. Prosodic variation for text-to-speech synthesis. Julia Hirschberg (AT&T Bell Labs., Rm. 2D-450, 600 Mountain Ave., Murray Hill, NJ 07974)

Appropriate intonational variation is critical for text-to-speech synthesis to convey appropriate meaning and improve naturalness. While in the past it has been thought that a full linguistic analysis is prerequisite to the generation of appropriate intonational features, in recent years considerable progress has been made in assigning prosodic variation in text-to-speech synthesis for unrestricted text, using simple and currently available techniques for text analysis. Techniques for varying pitch range, prominence, and phrasing based on such text analysis will be discussed. Also discussed will be techniques, such as phrasing and accent assignment procedures. Such automatic training allows quicker development of new prosodic variation procedures and also allows such procedures to be tailored to particular speaking styles or applications.

10:05

4aSP4. GEST: A computational model of speech production using dynamically defined articulatory gestures. Catherine P. Browman, Louis Goldstein, Elliot Saltzman, and Philip E. Rubin (Haskins Labs., 270 Crown St., New Haven, CT 06511)

A computational model of speech production that produces speech for English utterances, using dynamically defined articulatory gestures, will be described. The model is comprised of three submodels. The first part, the linguistic gestural model, generates a gestural score specifying the identity and relations of the gestures involved in the desired utterance. This gestural score is input to the second part, the task dynamic model [e.g., Saltzman and Munhall (1989)], which generates the movements of the various model speech articulators. These movements serve, in turn, as input to the third part, the vocal tract model [Rubin *et al.* (1981)], which determines the resulting area functions and acoustic signal. Attention will be focused on the integrated system and the first part, since the second and third parts of the model have been described elsewhere.

4aSP5. Segmental aspects of speech synthesis. Marian Macchi (Speech Systems Res. Dept., Bellcore, 445 South St., Morristown, NJ 07960)

A phonemic segment has different acoustic-phonetic realizations depending on many contextual factors, e.g., nearby phonemes and position in the syllable, word, and phrase. Appropriate acoustic variation in the phonemes is necessary for intelligible, natural-sounding synthetic speech. Two basic approaches to achieving these variations are: (1) articulatory synthesis, which models the human vocal apparatus, attempting to automatically account for the desired acoustic variability and (2) acoustic synthesis, which bypasses the articulatory level and operates on acoustic patterns directly, either by controlling phoneme-based format targets and transitions or by concatenating prerecorded units. In a phoneme-based system, whether articulatory or acoustic, the necessary acoustic variations for each phoneme are produced entirely by rules whose goal is to capture linguistic and articulatory regularities. However, knowledge of these rules is incomplete. The rationale for concatenative systems is that by recording and storing multiple variants of each phoneme, or units longer than phonemes, the units themselves incorporate some of the acoustic variation. Basic units for concatenation range in size and phonetic nature from phonemes or allophones, through dyads or diphones, polyphones, and demisyllables, to units covering polysyllables or words. This talk discusses interactions among the modeling approach, the basic unit, and the rules for combining units.

4aSP6. New directions for research on the perceptual evaluation of synthetic speech. David B. Pisoni (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

A review of previous research on the perceptual evaluation of synthetic speech produced by rule reveals that most studies have focused on behavioral measures of segmental intelligibility of isolated words. For the most part, except for a few isolated studies, research on sentence processing and comprehension of long passages of fluent speech has received very little attention. This is not surprising because these are areas of research in spoken language processing that are difficult to study without the investment of time and energy in developing new methodologies that go well beyond phoneme perception and isolated word recognition. Studies of prosody and the role of naturalness in intelligibility and comprehension are needed at this time to quantify the contribution of these sources of knowledge to spoken language processing. In this presentation, some of the recent studies from this laboratory that have attempted to measure these more complex processes will be reviewed. Studies using a sentence verification task and several on-line measures of comprehension will be described. Other methods need to be developed in the future to study intelligibility, comprehension, and retention of synthetic speech. [Work supported by NIH Research Grant No. DC00111-17.]

Contributed Papers

4aSP7. An algorithm for generating phrase boundaries for the automatic assignment of prosodic contours in a text-to-speech system for Spanish. Helen E. Karn (Dept. of Linguist., Georgetown Univ., Washington, DC 20057-1068)

Determining phonological and intonational phrase boundaries is an important step in synthesizing natural-sounding prosodic contours in a text-to-speech (TTS) system. An algorithm is presented here that generates phrase boundaries in Spanish texts. The basis for this algorithm is Liberman and Church's function group (*f*-group) parser for English [M. Y. Liberman and K. W. Church, "Text Analysis and Word Pronunciation in Text-to-Speech Synthesis," in *Advances in Speech Signal Processing*, edited by S. Furui and M. M. Sondhi (Marcel Dekker, New York, 1992), pp. 791-831]. The Spanish *f*-group parser produced satisfactory results for shorter sentences and sentences containing a fair amount of punctuation, but the parser tended to overgenerate phrase boundaries in longer sentences and sentences with little punctuation. In addition, the parser generated but did not distinguish different types of sentence and paragraph transitions. To refine the parser in these areas, selections from a Spanish text [E. Sabato, *El túnel* (Editorial Seix Barral, Barcelona, 1982)] were read by adult native Spanish speakers and analyzed for phrase boundaries as evidenced by significant changes in fundamental frequency, tempo, and/or pauses. The acoustic phonetic data and its role in the development of the phrase boundary algorithm are discussed.

4aSP8. Acoustics and synthesis of nasalization. Qiguang Lin, James Flanagan, and ChiWei Che (CAIP Ctr., Rutgers Univ., Piscataway, NJ 08855-1390)

In this paper acoustics and synthesis of nasalization is studied based on a comprehensive computer model of the vocal tract: TRACTTALK.

TRACTTALK simulates the vocal-tract system in the frequency domain and derives the time-domain equivalent to produce sound output. It incorporates all important components of the system and decomposes the transfer function into its zero and pole parts. Such a decomposition enables one to accurately estimate poles and zeros of a nasalized sound. First, temporal trajectories of poles/zeros are examined as a function of the velopharyngeal opening, the presence of the nasal sinuses, and other articulatory parameters. This is in follow-up to previous work [Flanagan, AT&T Bell Labs. internal report (1983)]. The attempt is to systematically characterize the pole/zero pattern of nasalization for improving the performance of formant tracking and feature labeling algorithms. Secondly, synthesis of nasalization is described using TRACTTALK. Listening experiments are conducted to assess the relationships among perceived nasality, inclusion of the nasal sinuses, and the degree of the velopharyngeal opening. It is found, e.g., that inclusion of the nasal sinuses results in stronger perceived nasality. While the literature reports differing views on the role of sinuses in nasality, these results are consistent with the findings of Maeda [Proc. ICASSP 2, 911-914 (1982)]. Synthetic speech generated from the vocal-tract system will be demonstrated.

4aSP9. A diagnostic text-to-speech intelligibility test. Ann K. Syrdal (AT&T Bell Labs., Rm. 3E-525, 101 Crawfords Corner Rd., Holmdel, NJ 07733) and Beki A. Sciacca (AT&T Bell Labs., Naperville, IL 60566 and Northwestern Univ., Evanston, IL 60208)

A wider sampling of speech segments and contexts is required for adequately evaluating text-to-speech output than is typically used in standard intelligibility tests of human speech. The test developed expanded upon both the Diagnostic Rhyme Test (DRT) [W. D. Voiers, Speech Tech. (Jan/Feb.), 30-39 (1983)] and a TTS intelligibility test developed by van Santen [J. P. H. van Santen, Comp. Speech Lang. 7, 49-100 (1993)]. The featural system used to construct the test ensured

that the most likely perceptual confusions were potential test items. One- and two-feature contrasts were tested in word-initial, word-medial, and word-final positions, in stressed and unstressed syllables. Vowel, consonant, and consonant cluster contrasts plus insertions and deletions were tested. Test items were presented in semantically anomalous sen-

tences (e.g., "Happy mimes dance the book.") and were controlled for familiarity and verb transitivity. Testing was automated and conducted interactively with individual subjects who received no feedback on their performance. Results indicate the test was a useful diagnostic tool as well as evaluation technique.

THURSDAY MORNING, 7 OCTOBER 1993

COLUMBINE ROOM, 7:55 TO 11:30 A.M.

Session 4aUW

Underwater Acoustics: Signal Processing I

Peter H. Dahl, Chair

Applied Research Laboratory, University of Washington, Seattle, Washington 98105

Chair's Introduction—7:55

Contributed Papers

8:00

4aUW1. Internal waves and matched-field processing. Darrell R. Jackson and Terry E. Ewart (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

The performance of matched-field processing is degraded due to randomness of the propagation medium. With a vertical array, this degradation takes the form of fragmentation and wandering of the peaks (mainlobe and sidelobes) in the range-depth ambiguity surface. The resulting errors in localization are characterized in terms of the rms processor output. Simulations and theory specialized to the cw Bartlett processor are used. First, the case of an ocean waveguide with a quadratic average sound-speed profile and vertically stationary sound-speed fluctuation statistics is examined. Next, approximations are introduced so that a relatively simple analytic model can be abstracted from the theory. This model is checked against Monte Carlo PE computations that avoid some of the simplifying approximations. The simple model contains scaling rules for processor performance as a function of frequency, array length, and medium vertical correlation length. In general, the effects of internal waves become more important as frequency increases, array length increases, medium correlation length decreases, and range increases. The processor is predicted to be most sensitive to internal-wave mismatch for sources that are in convergence zones. Finally, the analytical model is compared to Monte Carlo PE computations using oceanic realizations obtained from a realistic dynamic internal wave model.

8:15

4aUW2. Sector-focused matched-field inversion for enhanced environmental parameter estimation with reduced sensitivity to measurement errors. Howard A. Chandler, C. Feuillade, and G. B. Smith (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Sector-focused matched-field localization has been shown to effectively enhance range-depth resolution while remaining insensitive to environmental parameter mismatch. This becomes evident in parameter space ambiguity functions where the response is broad and therefore insensitive to mismatch. For source localization, this is a desirable result. However, for environmental inversion, greater resolution may be desired in parameter space while remaining insensitive to other measurement errors. This may be accomplished by including, in the search sector, replicas derived from a set of "neighboring" environmental parameters. The purpose of this study is to investigate the use of sector focusing to increase resolution in environmental parameter space while mitigating against measurement uncertainty. Results will be presented using the KRAKEN normal mode model to simulate a canonical shal-

low water environment. [Work sponsored by the Office of Naval Research, Program Element 61153N, with technical management provided by NRL-SSC.]

8:30

4aUW3. Trends in signal loss due to reflecting irregular bathymetry in matched-field processing. James F. Smith (Acoust. Div., Naval Res. Lab., Washington, DC 20375)

Simulations of the potential of matched-field tomography for inverting ocean sound-speed structure (Tolstoy and Diachok, 1991) have to date neglected effects of bottom interacting modes. A study to determine the effect of bathymetry in the thinly sedimented Pacific on matched-field processing has been conducted. A range- and depth-dependent sound-speed field was used. The acoustic field used as synthetic data "measured" at the array was calculated by the parabolic equation program FEPE (Collins, 1991). The program was shown to be highly accurate in matched-field studies against "exact" results produced by the normal modes program KRAKEN (Porter, 1991). The effects of irregular bathymetry and shifts in bottom sound speeds were studied through simulations conducted over extensive range and frequency intervals. The results showed an increase in signal array degradation with increasing range and frequency and reduced source localization. Effects of interaction with far and near sides of seamounts (relative to array position), and the overall trend of signal loss with range due to multiple interactions with the bottom will be discussed.

8:45

4aUW4. Optimal detection and source localization in an uncertain, shallow water environment. J. A. Shorey and L. W. Nolte (Dept. of Elec. Eng., Box 90291, Duke Univ., Durham, NC 27708-0291)

An optimal *a posteriori* probability approach to matched-field processing is implemented using the optimum uncertain field processor (OUFP) [A. M. Richardson and L. W. Nolte, *J. Acoust. Soc. Am.* **89**, 2280-2284 (1991)]. Combining Monte Carlo techniques with this processor results in a fast, efficient method for source localization or environmental parameter estimation when the environmental parameter search space is large. Detection and localization performance is presented using the ROC for detection and the LROC (localization receiver operating characteristic) for source localization. Specific results are presented for a shallow water environment as a function of signal-to-noise ratio and environmental uncertainty. [Research supported by ONR under Contract No. N00014-91-J-1448.]

4aUW5. Environmental source tracking using measured replica fields. W. A. Kuperman,³⁾ Michael D. Collins, John S. Perkins, Laurie T. Fialkowski, Timothy L. Krout (Naval Res. Lab., Washington, DC 20375), Lindsay Hall, Ralph Marrett (Defence Scientific Establishment, Auckland, New Zealand), Lesley J. Kelly, Ashley Larsson (Defence Science & Technology Organisation, Salisbury, SA, Australia), and John A. Fawcett (Defence Research Establishment Pacific, Victoria, BC, Canada)

Preliminary results will be presented for TESPEX (Test of Environmental Signal Processing Experiment), which was performed in May 1993 off the east coast of New Zealand in a region of three-dimensional bathymetry variations. This complex environment was exploited to minimize ambiguity in environmental source tracking with a single receiver [Collins *et al.*, *J. Acoust. Soc. Am.* **90**, 2366 (1991)]. To overcome limited knowledge of environmental parameters, the acoustic field was measured by a fixed array of receivers while a ship towing a source swept over a sector. The receivers were linked to a recently developed satellite telemetry buoy that transmitted time series to a centralized computer facility for real time analysis. Expensive ship time was traded off for cheap computation time by interpolating the acoustic field using a WKB representation that permits a sparse sampling in azimuth. The main computational task for the data basing involves solving a nonlinear optimization problem for the WKB amplitude and phase functions, which vary slowly with range and azimuth. TESPEX data have been used to perform environmental source tracking using replicas constructed from the acoustic field data base. ³⁾Present address: Scripps Institution of Oceanography, La Jolla, CA 92093.

9:15

4aUW6. Oceanic tomography using model-based matched-filter processing of wideband acoustic signals. J.-P. Hermand (SACLANT Undersea Res. Ctr., Viale San Bartolomeo, 400, I-19138 La Spezia, Italy) and W. I. Roderick (Naval Undersea Warfare Ctr., Newport, RI 02841)

As originally conceived, ocean acoustic tomography relies on determining the travel times of pulses that propagate along identifiable multipaths between pairs of transducers. In principle, other properties of acoustic propagation such as amplitude and phase over a broad frequency band can be used to infer environmental parameters. These parameters include range-dependent sound-speed profiles and geoacoustic parameters of the sea bottom. Recent experimental work has demonstrated that the distortion of time-dispersed wideband signals can be compensated effectively by using a model-based matched filter (MBMF). A multi-channel MBMF receiver that incorporates the modeled Green's function of the medium, determined the correct range and depth of the source and receiver. In this paper, the possibility of applying model-based matched-filter processing to the tomographic inversion problem was investigated. The acoustic data are low-frequency, large time-bandwidth product, linear-frequency-modulated signals transmitted through a range-independent waveguide at a deep water site west of Sardinia. Compared to time-of-flight tomography, the amplitude and phase distortion undergone by the propagated waveforms were fully exploited to reconstruct, in part, the range-averaged sound-speed profile between source and receiver. This was achieved by searching for the set of sound-speed profile parameters that maximized the gain at the output of a multi-channel MBMF receiver.

9:30-9:45 Break

9:45

4aUW7. The tomographic inverse beamforming (TIBF) solution—A new matched-field algorithm. James H. Wilson³⁾ (Dept. of Oceanogr., Naval Post Graduate School, Monterey, CA 93943)

The recent at-sea, real-time signal detection and tracking performance of the plane-wave solution to the inverse beam forming (IBF) integral equation (Wilson, 1983; Nuttall and Wilson, 1990) has shown

significant (5 to 12 dB) gains compared to operational sonar systems and other adaptive processing methods. The Fourier integral method (FIM) (Nuttall and Wilson, 1990) was thought to be the "spikiest" solution to the IBF integral equation, that is also *linear* in the covariance matrix. A standard inverse technique (Backus and Gilbert, 1968) used in tomography has recently been applied to the IBF integral equation, and two new theoretical results have been obtained. First, the plane-wave solution, called the least-squares Wilson integral method (LSWIM), agrees with FIM only at the array design frequency, and is spikier or more "delta function like" than FIM below array design frequency. Second, the non-plane-wave or matched-field solution was obtained by allowing the measured data vector (covariance matrix) in the Backus-Gilbert inverse method to have *two* discrete indices instead of the usual *one* index. This work was performed during the analysis of Outpost SUNRISE data for the purpose of enhancing sonar detection and tracking performance, but has also a more general application in acoustical oceanography. [Work supported by AEAS.] ³⁾On temporary leave from Neptune Sciences, Inc.

10:00

4aUW8. Extraction of both bottom backscattering strength and reflection loss by inversion of reverberation measurements. Sicco D. Kamminga,¹⁾ Dale D. Ellis, and Peter Gerstoft (SACLANT Undersea Research Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy)

A method is described to simultaneously extract the bottom backscattering strength and bottom reflection loss as a function of grazing angle from measurements of monostatic reverberation. The least-squares difference between a parametrized model and data is minimized with respect to the parameters using generic algorithms or simulated annealing. The reverberation data are from broadband sources and omnidirectional receivers deployed at a flat-bottomed deep-water area in the Mediterranean. The reverberation model is simple: the propagation is described by straight-line ray paths (although the effective angles are corrected for the sound-speed profile); and parametrized functions are used for the bottom scattering strength and reflection loss. The parameters can also include the water depth, sound speed and gradient, source level, and pulse length. Where they can be compared, the parameters resulting from the inversion show good agreement with the measured ones, leading confidence to the procedure. As well, when the resulting parameters and actual sound-speed profile are input to the generic sonar model, the calculated and measured reverberations are in good agreement. ¹⁾Currently at Rijkswaterstaat, P.O. Box 3006, 2280MH Rijswijk, The Netherlands.

10:15

4aUW9. Track-before-detect matched-field processing. Paul A. Baxley (Ocean and Atmospheric Sci. Div., Code 541, NRAD, NCCOSC, San Diego, CA 92152-5000), Robert Bruce Williams (NRAD, NCCOSC, San Diego, CA 92152-5000), and William H. Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA 92152-6400)

Matched-field processing (MFP) is well-suited for tracking applications because of its inherent source localization capability, as demonstrated by Wilmut *et al.* [*J. Acoust. Soc. Am.* **93**, 2374 (A) (1993)]. If exact track determination is not an objective, improved detection at a low signal-to-noise ratio may be obtained using a simple track-before-detect technique that averages range-shifted MFP ambiguity-surface snapshots, with the range shift being determined by an assumed range rate. After searching through all candidate range rates, the signal function is approximately reconstructed in the average surface corresponding to the correct range rate. If peak-to-sidelobe relationships in the signal function vary little between snapshots, both the sidelobes and the main peak are maximized in the average surface for the correct range rate; otherwise, only the main peak is maximized. In either case, this maximization provides a detection clue along with an estimate of range rate. This detection enhancement at low signal-to-noise ratios is demonstrated via simulations for a 48-element full-water-column vertical array in a thickly sedimented shallow-water environment (500-m water depth) typical of the southeastern Mediterranean sea. A 50-Hz source

moving at constant range rate and source depth was simulated using a normal mode program, while noise samples were obtained using a PE-based shipping noise model and the HITS II shipping distribution database.

10:30

4aUW10. Acoustic signatures of imploding underwater light bulbs. William J. Marshall, Jr. (BBN Systems & Technol., Union Station, New London, CT 06320)

Imploding light bulbs are sometimes used as underwater sound sources because they produce impulsive signals similar to small explosions, yet are cheaper and more convenient to procure, store, and use. On a recent sea trial, light bulb sources were used to measure the straightness of a bottom-mounted hydrophone array. For this purpose the sharp leading edge was the most important characteristic of the acoustic signature, however the test provided a good opportunity to record and study other signature features as well. Results for a moderate number of common household light bulbs are given. Data include statistics on crush depths, measured acoustic waveforms, and energy spectral levels, and correspondence between these observations and a simple theory of sound generation by imploding gas bubbles. This information should be useful to others planning acoustical tests using such sources. [Work sponsored by the Advanced Research Projects Agency.]

10:45

4aUW11. Insight into the learning process of neural networks for classifying acoustic signatures. Fred C. DeMetz, Sr. (Allied Signal Ocean Systems, Inc., 15825 Roxford St., Sylmar, CA 91342) and Lowell W. Brooks (Technol. Service Corp., P.O. Box 1120, Salida, CO 81201)

Feedforward networks employing the backward propagating delta rule for error correction were tested utilizing simulated target signatures and noise to provide insight into the network learning process. Network training histories and weight evolutions were studied for alternating signal and noise input vectors for two network architectures. Contour plots of the input-to-hidden layer weights clearly indicate the relationship between the evolving features of the network weights as they respond to the input signatures during the learning process. Singular value decomposition of the input-to-hidden layer transfer matrix provides in-

sight into the similarities of a trained neural network and a classical matched filter.

11:00

4aUW12. Towed array processing as a spatial Kalman filter problem. E. J. Sullivan (Naval Undersea Warfare Ctr., Code 103, Newport, RI 02841) and J. V. Candy (Lawrence Livermore Natl. Lab., Livermore, CA 94550)

A moving towed array essentially constitutes a system intended to provide a series of measurements that are sequential in space. This type of process fits quite naturally into an iterative measurement process that can be realized as a spatial predictor-corrector form of a Kalman filter. It is shown how this concept is formulated and constitutes a generalization of the overlap-correlator synthetic aperture algorithm [E. J. Sullivan and S. Stergiopoulos, *J. Acoust. Soc. Am.* **86**, 158-171 (1989)]. Examples will be given that demonstrate how this scheme constitutes a self-consistent synthetic aperture algorithm capable of determining estimates of the bearings of multiple sources, both correlated or uncorrelated. It will also be shown how the method provides coherent synthetic aperture processing over integration times that are significantly longer than the coherence time of the source signals.

11:15

4aUW13. Scattering function description of a shallow water channel. Peter G. Cable (BBN Systems and Technol., Union Station, New London, CT 06320-6147)

The shallow water channel is considered as a random, time-varying linear filter. The sources of temporal variation of channel properties considered are source and receiver motion and the motion of the surface (that is, wind waves and swell). Using an energy flux description for acoustic propagation, after Smith, Weston, and Brekhovskikh, and employing Smith's theory of coherence in multimode propagation, an explicit formulation of the scattering function for a shallow water channel has been obtained. The scattering function description seems suited to a mid or high frequency (5-20 kHz) description of the shallow water channel. The conditions necessary for the scattering function description, namely that the channel be wide-sense stationary with uncorrelated scatterers (WSSUS), are derived and discussed. Specific examples of the scattering function for different channels will be presented and discussed.

THURSDAY MORNING, 7 OCTOBER 1993

SAVOY ROOM, 8:30 A.M.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

U.S. Technical Advisory Group for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protection

D. L. Johnson, Chairman S12

EG&G Special Projects, Albuquerque Operations, Albuquerque, New Mexico 87119-9024

H. E. von Gierke, Chairman, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
1325 Meadow Lane, Yellow Springs, Ohio 45387

Elliott H. Berger, Chairman, U.S. Technical Advisory Group (TAG) for ISO/TC 94/SC12, Hearing Protection
Cabot Safety Corporation, 7911 Zionsville Road, Indianapolis, Indiana 46268-1657

Standards Committee S12 on Noise. Working group chairs will report on their progress for the production of noise standards. The interaction with ISO/TC 43/SC1 and ISO/TC 94/SC12 activities will also be discussed, with reference to the international standards under preparation. The Chairs of the respective U.S. Technical Advisory Groups (H. E. von Gierke and E. H. Berger) will report on current activities of these International Technical Subcommittees under ISO and results of the last meeting of ISO/TC 43/SC1, which took place in Oslo, Norway from 31 May-4 June 1993.

Scope of S12: Standards, specifications, and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation, and control; including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

Session 4pAA

Architectural Acoustics: Room Acoustics and Sound Isolation

Edward R. McCue, Chair

Kirkegaard and Associates, 954 Pearl Street, Boulder, Colorado 80302

Chair's Introduction—1:20

Invited Paper

1:25

4pAA1. Subjective and objective parameters of sound within fabric structures. Andrei N. Chesnokov (Lab. of Architectural Acoust., Res. Inst. of Bldg. Phys., Locomotivny pr. 21, Moscow 127238, Russia)

An investigation of acoustics in 15 circuses with fabric roofs has been carried out. Objective acoustical parameters (reverberation time, clarity C_{80} , and others) have been measured and questionnaires from the audience in the circuses have been obtained. Features of circuses acoustics are the decrease in reverberation times at low frequencies, echo (in halls with impervious fabric roofs), and high levels of background noise. The background noise levels in circuses before a performance were equal $L_A=46-56$ dBA. The questionnaire contained subjective acoustical attributes: Timbre, balance, reverberance, spaciousness, echo, loudness, clearness, audibility of background noise, background noise affect on speech legibility, and others. Average mark estimates and results of measured noise level in halls (L_A) have been subjected to factor analysis. Three factors have been extracted that can be interpreted as (1) overall acoustical impression of the music sound in a circus, (2) disturbing influence of the background noise, and (3) clarity of music and speech sounding.

Contributed Papers

2:05

4pAA2. A comparison of three concert halls using impulse response spectrograms. Mendel Kleiner, Patrik Falk, and Björn Olsson (Dept. of Appl. Acoust., Chalmers Univ. of Technol., S-41296 Gothenburg, Sweden)

Three concert halls, Symphony Hall (Boston), Concert Hall at Otis A. Singletary Center for the Arts (Univ. Kentucky, Lexington), and Konserthuset (Gothenburg, Sweden) have been investigated. The floor plans and sections of these halls are fundamentally different as are the scattering properties of the walls and ceilings of the halls. Impulse responses were recorded in a number of positions in each hall. These impulse responses were then analyzed using FFT and Wavelet spectrograms to study to what extent the obvious geometrical differences could be traced in the impulse responses. The spatial distribution of the sound in the halls was studied using a "running" short-time interaural cross-correlation technique based on the impulse responses recorded using a dummy-head. Results show that the wavelet spectrogram yields graphs which may be slightly easier to analyze than conventional FFT spectrograms. The expected differences due to geometry are visible in the spectrograms. The short-time interaural cross-correlation graphs add information which also agrees with expectations.

2:25

4pAA3. The Great Hall at Cooper Union: A study of its acoustical problems. Daniel R. Raichel, Tanya Dragan, and Winston Wilkerson (Albert Nerken School of Eng., The Cooper Union for the Advancement of Science and Art, 51 Astor Place, New York, NY 10003)

The Great Hall, a place for public assembly in the lower level of the Foundation Building at the Cooper Union in New York City, is steeped in history. Constructed in 1859, it was the site where Abraham Lincoln have his "Right Makes Might" speech that catapulted him to the Republican presidential nomination. Many prominent speakers, including Mark Twain, feminist Victoria Woodhull, Theodore Roosevelt, Wood-

row Wilson, labor leader Samuel Gompers, and very recently President Clinton, graced the podium, a tradition that continues to this day; and the hall also accommodates musical events and theatrical productions. The Great Hall underwent reconstruction nearly 20 years ago, but recent concerns about its acoustical properties led to the authors' measurements of the reverberation times in different parts of the 900-seat auditorium. Reverberation times exceeding 3 s in the 125- to 1000-Hz range point up the existence of excessive echoes that made speech comprehension difficult for many members of the audience. Solutions are being developed to lower the reverberation times and to provide flexibility for meeting the requirements of different types of events, with due regard for the landmark status of the Foundation Building.

2:45

4pAA4. A nonlinear regression approach to the evaluation of reverberation times using Schroeder's integrated impulse response method. Ning Xiang (HEAD Acoust., Kaiserstr. 100, D-5120 Herzogenrath 3, Germany), W. Ahnert, and R. Feistel (ADA Acoust. Design Ahnert, Berlin, Germany)

Reverberation decay curves can be obtained by backward integration of room impulse responses [M. R. Schroeder, *J. Acoust. Soc. Am.* 37, 409-412 (1965)]. The evaluation of reverberation times is often achieved by a regression line fitting the reverberation decay curves. However, the successful application of this method requires either a careful choice of the integration limit or estimation of the mean-square value of background noise where background noise is present in the room impulse responses to be evaluated. In the present paper, an alternative method for evaluating reverberation times from Schroeder's decay curves using a nonlinear iterative regression approach is proposed. The regression process is based on a nonlinear curve model using the generalized least-square error principle rather than a linear model as used in the linear regression. The present paper will describe the principle of this approach and discuss the advantages and disadvantages involved. Comparison of results obtained using this approach and alternative methods will also be presented.

3:05

4pAA5. Experiencing the aural environment of buildings. Gary W. Siebin and Martin Gold (Dept. of Architecture, 231 ARCH, Univ. of Florida, Gainesville, FL 32611-2004)

A simple audio system was developed to allow architecture students to simulate the aural environment of various rooms. A basic set of exercises provide a conceptual and aural understanding of the room impulse response method for evaluating subjective acoustic qualities of rooms. The focus of the exercises is to provide a structure for the experiential understanding and manipulation of acoustics of buildings in applied design situations. Aural demonstrations illustrating how changes in the impulse response such as the loudness, direction, and number of early reflections as well as the subsequent reverberation affect the perceived acoustical qualities of speech and music are shown for various environments. The exercises are implemented using anechoic music recordings played via loudspeakers or headphones. The impulse responses are simulated using digital delay and reverberator units. Once they become familiar with the equipment, students can actually manipulate listening conditions easily for an endless opportunity of listening experiences.

3:25-3:40 Break

3:40

4pAA6. Spectral properties of transmission in a simulated room. William J. Strong and Sining Li (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602)

Transmission of sound in a large room was simulated by means of an image method incorporating artificial diffusion. The simulation model was validated by comparing the standard deviations of its transmission spectra to theoretical and experimental results obtained by others. Several combinations of source and receiver positions were simulated in order to assess their effects on spectral variability. A single source and single receiver arrangement resulted in the greatest variability as expected. Other combinations, including single-source/two-receiver (to represent two ears), two-source (to represent spatial extent of source)/two-receiver, and five-source/five-receiver, were used to assess their effects. The effects of frequency modulation and critical bands of the ear were also simulated. In an attempt to relate simulation results to stability in perception, a limit of 6-dB difference in relative harmonic levels between received tones and source tones was chosen as the criterion. On the basis of this criterion, a single-source and two-receiver arrangement incorporating critical band effects and frequency modulation was found adequate to provide stable perception.

4:00

4pAA7. Music as noise: Sound transmission and unbiased annoyance. David A. Nelson (Acoust. Systems Acoust. Res. Facility, P. O. Box 3610, Austin, TX 78764)

Music is often perceived as noise pollution, especially when it invades one's solitude or one's own music making. This is generally a major issue in multiunit residential buildings and in music facilities for rehearsal, broadcast, and recording. However, current noise control methods of design and classification do not address the ongoing trend in

popular music toward strong bass content. Analysis of broadcast music signals using unbiased annoyance [E. Zwicker, "On the dependence of unbiased annoyance on loudness," Proc. Internoise 89, pp. 809-814] indicates that (i) contemporary music can be adequately modelled as pink noise from 50 to 4000 Hz, (ii) after passing through a partition the 63-Hz octave band dominates loudness and unbiased annoyance, (iii) existing single-number ratings (such as STC) cannot identify structures with critical low-frequency resonances, thus potentially overestimating performance against this type of noise. Ramifications for noise control practice and effective single-number ratings are explored.

4:20

4pAA8. Time response analysis of sound fields in rooms and the temperature condition of the air. Hidemaro Shimoda (Inst. of Technol., Shimizu Corp., Etchujima 3-4-17 Koto-ku, Tokyo, 135 Japan), Tatsuya Kashiwa, Norinobu Yoshida, and Ichiro Fukai (Faculty of Eng., Hokkaido Univ., N-13 W-8 Kita-ku, Sapporo, 060 Japan)

Transient analysis in the time domain is very important to determine acoustical properties in rooms. In this paper, the transmission line network is used as a simulation code named Bergeron's method [J. Acoust. Soc. Am. Suppl. 1 84, S64 (1988)]. Time responses in the rectangular scale model excited with pure tones were compared to analytical results. In the study, the medium condition as a temperature of the air turned out to be very sensitive to simulated time responses in sound fields in rooms. Even though it is known that the sound speed varies with a change in the temperature of the air, the standard sound speed corresponding to the standard temperature of the air, such as 15 °C in ordinary sound field simulations, is often used. Analytical time responses, however, as compared with experimental responses in the model, were found to be very different from each other in about 1% change of the sound speed.

4:40

4pAA9. Necessary accuracy for early reflections in sound field auralization. Toshiyuki Okano (Takenaka Res. and Development Lab., 2-5-14 Minamisuna, Koto-Ku, Tokyo, 136 Japan)

Necessary accuracy for early reflection levels was investigated in order to derive criteria to verify the accuracy of a sound field simulator. Early reflections are the most important in controlling the subjective impression of a room's sound field. The accuracy of the sound field simulator depends primarily on the accuracy of early reflections. The just-noticeable level difference (jnld) of early reflections defines the maximum of allowable level difference between real and simulated reflections. Also, the masked threshold level (MTL) of early reflections defines the level of the smallest reflections which a simulator should generate. However, jnld and MTL of early reflections depend on the sound field's configuration. It is desirable that criteria are independent of the configuration of the sound field. The jnld and MTL were measured for lateral and ceiling reflections in various types of rooms with simulated sound fields. Measured jnlDs and MTLs were converted into the difference in objective parameters (LE, IACCE, C80, etc.), which correspond to subjective impressions caused by early reflections. Criteria for necessary accuracy were pursued in the domain of rooms with objective acoustic parameters.

Session 4pAB

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Effects of Noise on Marine Mammals II

Charles I. Malme, Chair

Bolt, Beranek and Newman, Inc., 70 Fawcett Street, Cambridge, Massachusetts 02138

Chair's Introduction—12:55

Invited Papers

1:00

4pAB1. Variability in behavioral reaction thresholds of bowhead whales to man-made underwater sounds. W. John Richardson (LGL Ltd., Environmental Research Associates, 22 Fisher St., P.O.B. 280, King City, ON L0G 1K0, Canada) and Charles R. Greene, Jr. (Greeneridge Sciences, Inc., Santa Barbara, CA 93110)

Reactions of bowhead whales, *Balaena mysticetus*, to actual and simulated oil industry activities were studied in the Beaufort Sea during spring, summer, or autumn from 1980 to 1991. Received levels of oil industry noise that elicited behavioral reactions varied widely, depending partly on the source and characteristics of noise, the activity of whales when exposed, and the physical situation (all levels in dB re: 1 μ Pa, 20–1000-Hz band). Reactions to increasing noise levels from approaching boats occurred at received levels as low as 90 dB. In summer, roughly half the bowheads showed avoidance when the received level of steady drillship or dredge noise was about 115 or 20-dB above ambient. Results in spring were similar, but some bowheads tolerated received levels of drilling sound up to 135+ dB if the only available migration route through ice required close approach to a sound projector. In contrast, repetitive pulses from airgun arrays elicited avoidance only at received levels exceeding 150–180 dB (although weaker pulses may elicit subtle behavioral changes). Besides the variability among sound types and situations, there is much whale-to-whale variation in any one situation. No single received level criterion is appropriate to all situations. [Work supported by Minerals Management Service.]

1:15

4pAB2. Responses of wintering humpback whales to vessel traffic. Gordon B. Bauer (Div. of Social Sci., New College, Univ. of South Florida, 5400 North Tamiami Trail, Sarasota, FL 34243-2197), Joseph R. Mobley (Univ. of Hawaii—West Oahu, Pearl City, HI 96782), and Louis M. Herman (Univ. of Hawaii, Honolulu, HI 96822)

Responses of humpback whales to vessel traffic were monitored over two winter seasons during 1983–1984 in Maui, Hawaii. A variety of vessel characteristics including vessel numbers, speed, and proximity were associated with changes in whale behaviors, including swimming speed, respiration, and social behaviors. Smaller pods and pods with a calf were more affected than larger pods. A case study indicated that a calf could be sensitized by the passby of a large vessel, so that it subsequently breached in response to noise from a smaller boat engine which had not previously elicited any behavior change. These findings in conjunction with similar results from summering humpbacks in Alaska indicated disturbance of humpback whales at both ends of their range. Although substantial short-term effects were noted, long-term negative consequences are not apparent. Recent aerial surveys of the Hawaiian Islands indicate substantial increases in the number of humpback whales.

1:30

4pAB3. Responses of humpback whales to playback of natural and artificial sounds in Hawaii. Adam S. Frankel and Louis M. Herman (Dept. of Oceanogr. and Psychol., Univ. of Hawaii at Manoa, 1000 Pope Rd., Honolulu, HI 96822)

Humpback whales were exposed to the playback of humpback song, social sounds, Alaskan feeding call, artificially synthesized sounds, and blank tape control with a source level of 162 dB re: 1 μ PA. Whales were observed and tracked from shore. A 20-min baseline observation preceded the 20-min sound presentation. The J-11 transducer was deployed at 7-m depth in water from 20 to 200-m deep. Acoustic velocity profiles conducted nearby indicated a uniform acoustic velocity field. The strongest reaction observed was the rapid approach response where whales changed course and approached the boat. The percentage of such responses varied with sound condition. Rapid approach responses were observed from whales up to a 1.5-km distance. Received levels can be approximated from the range between transducer and whale. The minimum received level that produced a strong reaction was probably between 100–115 dB, for 20 and 15 log *R* transmission loss models. These estimates could be improved as some playback experiments were recorded by a second vessel.

1:45

4pAB4. Responses of humpback whales to sonar sounds. Hilary L. Maybaum (Ogden Environmental, 680 Iwilei Rd., Ste. 660, Honolulu, HI 96817 and Dept. of Oceanogr., University of Hawaii, 1000 Pope Rd., Honolulu, HI 96822)

Controlled sound playback experiments were used to assess effects of a low-frequency sonar system on humpback whales, *Megaptera novaeangliae*, in Hawaiian waters. Focal pods were presented with sounds of a 3.3-kHz sonar pulse, a sonar frequency

sweep from 3.1 to 3.6 kHz, or a control (blank) tape. Behavior, movement, and underwater vocalizations were monitored and compared with baseline periods. While the two types of sonar signals differed in their effects on the whales, both elicited avoidance behaviors. Humpbacks responded to the pulse by increasing their distance from the sound source. The strength of this effect varied directly with time. Responses to the frequency sweep primarily consisted of increased swimming speeds and track linearity. The latter was a direct function of increasing sound intensity. Overall, the sounds did not strongly or consistently affect the whales' dive cycles or vocalizations. Observed avoidance reactions may have resulted from possible resemblance between the sonar signals and natural sounds in humpbacks' environment that are associated with biological threats or warnings.

2:00

4pAB5. Vocalizations of blue and fin whales during a midocean ridge airgun experiment. Mark A. McDonald, John A. Hildebrand, Spahr Webb, LeRoy Dorman (Scripps Inst. of Oceanogr., Univ. of California, La Jolla, CA 92093-0205), and Christopher G. Fox (OSU Hatfield Marine Sci. Ctr., Newport, OR 97365-5258)

Numerous seismic experiments are conducted each year in the deep oceans to study the nature of oceanic crust and to map the source of seismic signals associated with small earthquakes or volcanic activity. Whale vocalizations of the type associated with blue and fin whales are often recorded on the arrays of seafloor seismometers and hydrophones used for these experiments. These whale vocalizations are characterized from one such experiment conducted about 500-km offshore from Astoria, Oregon in August of 1990. The travel time differences and signal amplitudes from both direct and multipath arrivals across the seafloor seismometer array are used to locate the whales and predict the level of ship and airgun noise at the whale. Whale vocalizations were recorded during airgun operations and these vocalization patterns are compared to patterns recorded during times of relative quiet from the 12 days of data gathered in this experiment. Transient oceanic sound levels from transform fault earthquakes and seafloor volcanic activity are often louder than those produced by airguns.

2:15

4pAB6. Underwater earthquakes noise levels and its possible effect on marine mammals. Clyde E. Nishimura (Naval Res. Lab., Code 7420, Washington, DC 20375) and Christopher W. Clark (Cornell Univ., Ithaca, NY 14850)

Earthquakes in oceanic regions commonly generate acoustic signals known as T-phases which are similar but not identical, to man-made explosions. As earthquake-producing areas are also regions where marine mammals congregate (e.g., Aleutians and Caribbean margin), knowledge of the characteristics of T-phases may provide some additional information on the possible effect of noise on these animals. T-phases are generally characterized by acoustic energy below 100 Hz with most of its energy in the 10- to 30-Hz range. The duration of the T-phase is, to the first order, linearly related to the source earthquakes magnitude; durations of several minutes are common. The T-phase source signal level, which can exceed 200 dB *re*: 1 μ Pa for a magnitude 4–5 earthquake, is a complex function of the source magnitude, focal depth, and the complexity of the bathymetry at the water–rock radiation area. This radiation area is not a point source and can extend over a radius of several tens of km. As part of the Whales '93 program, analysis has begun as to whether there is any correlation between the occurrence of earthquakes and observable changes in the acoustic signature form, and the positioning of large cetaceans.

2:30

4pAB7. The reaction of humpback whales to underwater explosions: Orientation, movements, and behavior. Jon Lien, Sean Todd, Peter Stevick, Fernanda Marques (Whale Res. Group, Memorial Univ., St. John's, NF A1B 3X9, Canada), and Darlene Ketten (Harvard Med. School, Boston, MA 02114)

In 1992, local fishermen reported unusually high net collision rates by humpback whales in Bull Arm, Trinity Bay, Newfoundland (47° 45'N, 53° 50'W), an area of underwater industrial activity. As part of a study to investigate this phenomenon [see also Ketten *et al.*, this meeting], levels and types of noise—including underwater explosions—were sampled. The location and movement of a small group of humpbacks (71 individuals identified over a 19-day period) resident in Bull Arm were monitored; when possible, behavior of individuals was recorded directly. CTD profiles and bait abundance were also noted. Explosions were of high amplitude and low frequency. Measured at 1 mile from source, levels typically reached 150 dB (*re*: 1 μ Pa at 1 m, at 350 Hz). Following explosions, residency time and location of individual humpbacks did not change. When individuals could be observed directly, no behavioral reaction to explosions (sudden dives, abrupt movements) were seen. Although not statistically significant, more animals were sighted and resighting rates were higher in the explosion area than in other parts of the bay. However, two animals recollided with fishing gear—such reports of successive entrapments are rare.

2:45–3:00 Break

3:00

4pAB8. Blast injury in humpback whale ears: Evidence and implications. D. R. Ketten (Dept. of Otolaryngol., Harvard Med. School, MEEI, 243 Charles St., Boston, MA 02114), J. Lien, and S. Todd (Memorial Univ., St. John's, NF A1B 3X9, Canada)

To date, there is no published report of effects on marine mammal hearing from underwater explosions. External injuries consistent with inner ear damage have been found in dolphins subjected to Class C explosives, but often little change is seen in surface animal behavior near blast areas [Richardson *et al.*, OCS MMS/90-0093 (1991)]. In this study, temporal bones from two humpback whales, which died following a 5000-kg explosion in Trinity Bay, Newfoundland [Lien *et al.*, *J. Acoust. Soc. Am.* **94**, 1849(A) (1993)], were harvested, preserved in formalin, scanned with 1-mm-high resolution spiral CT, decalcified, and sectioned at 20 μ . Evidence of mechanical trauma was found in all four ears: Round window rupture, ossicular chain disruption, sero-sanguinous effusion of peribullar spaces, and dissection of the middle ear mucosa with pooled sera. In one animal, there were bilateral periotic fractures. These observations are consistent with blast injury reports in humans, particularly with damage to

victims near the source who sustained massive, precipitous increases in cerebrospinal fluid pressure. There was no evidence that the pathologies found in these whales resulted from repeated barotrauma or chronic infection, and no similar abnormalities were found in control ears from humpbacks not exposed to blasts. While the results show whales, like other mammals, are subject to severe blast trauma, it remains unclear whether lower level stimuli induce temporary and/or acute threshold shifts in marine mammals. [Work supported by ONR Grant. No. N00014-92-J-4000.]

3:15

4pAB9. Prediction of potential disturbance of baleen whales by low-frequency acoustic transients. Charles I. Malme (25 Rockwood Rd., Hingham, MA 02043)

High-energy, low-frequency sound sources are useful tools for geophysical surveying, submarine detection, and long distance acoustic tomography. These sources produce impulsive, narrow-band and swept-tonal signals at high levels in the oceanic environment. This study was made to estimate the received level-time duration characteristics of acoustic transients that can potentially influence baleen whale behavior, the species of particular concern. Findings of acoustic disturbance studies show that continuous sound levels >120 dB *re*: $1 \mu\text{Pa}$ produces $>50\%$ avoidance by gray and bowhead whales. However, for impulsive airgun sounds of duration <0.5 s, effective pulse levels 30 to 50 dB higher are required to produce 50% avoidance for the same species. Little information is available on whale response to intermediate sound durations representative of some sonar and tomography source operations. Consequently, the literature on human response to acoustic transients was examined for response prediction methodologies suitable for application to whale acoustic response. Similarities were found that support the application of a modified equivalent level (L_{eq}) metric. The exposure level-time duration characteristics obtained from this analysis are preliminary estimates. The predictions should be tested using data obtained in the oceanic environment using representative sources, signals, and whales. [Work supported by the U.S. Navy.]

3:30

4pAB10. Recording, analysis, and verification of low-frequency sounds produced by Arctic offshore drilling equipment. Kenneth D. Brewer and John D. Hall (Coastal & Offshore Pacific Corp., P. O. Box 31554, Walnut Creek, CA 94598)

Digital recordings of underwater sounds produced by anchored and bottom-founded Arctic offshore drilling equipment (drilling rigs) were made during drilling and idle conditions in open water and under sea ice. Sound production was verified by recording the frequencies produced by specific pieces of equipment aboard the rigs using a single axis accelerometer. It was determined that most of the sound energy produced by both the anchored and bottom-founded drilling equipment was in the frequency band from 1 Hz to 1 kHz. The drill turntable and drive equipment on the bottom-founded drilling rig produced sounds as low as 1.8 Hz. The broadband [20 Hz to 20 kHz source level (dB *re*: $1 \mu\text{Pa}^2@1 \text{ m}$)] of the bottom-founded drilling rig (146.2 dB) was about 30 dB less than the broadband source level of the anchored drilling rig (178.6 dB). Acoustic transmission loss in the inshore areas of the Alaskan Beaufort Sea, where water depths are less than 100 m, was found to be slightly greater than cylindrical spreading (-11.0 to $-13.0 \log R$ dB) at the two sites about 350 km apart. [Work supported by ARCO Alaska, Inc.]

3:45

4pAB11. Variation in received level from man-made low-frequency underwater noise sources as a function of diving animal depth. William T. Ellison, Karen S. Weixel (Marine Acoust., Inc., 14 Pelham St., Newport, RI 02840), and Christopher W. Clark (Cornell Univ., Ithaca, NY 14850)

Interest in the effects of low-frequency (<1 kHz) man-made noise on marine wildlife highlights the need for accurate knowledge of the spatial distribution of noise levels within a given wildlife habitat. These levels can vary significantly, particularly with respect to depth within the water column. For diving animals, the ability to measure or predict this variation with depth is a necessary factor in assessing the net impact of that noise; i.e., one needs to perform a convolution of diving patterns (depth as a function of time) with the noise pattern as a function of time and depth. Recent advances in range dependent acoustic propagation modeling allow for the ability to predict with some accuracy the transmission loss from a known source of noise as a function of source characteristics (directivity, sound spectrum, location, and depth) to any given location in a hypothetical habitat. Several examples are presented that illustrate how these models might be used to evaluate the net impact of a passing noise source on pelagic whale species. [Work supported by ONR.]

4:00

4pAB12. Modeling noise interference with animal communication. Frank T. Awbrey (Biology Dept., San Diego State Univ., San Diego, CA 92182-0057)

Regulators are proposing to set simplistic limits for anthropogenic noise in natural environments. These proposed limits specify only SPL, without regard to the method of measurement, frequency or time weighting, normal ambient levels, or spectra. In some cases such limits may be inadequate, but in others they may be much too severe, resulting in unrealistic restrictions on human activities, including scientific investigations, without real benefit to the animals. Realistic limitations on noise level should incorporate information on the time history and spectrum levels of the noise and the animals' communication signals, with estimates of hearing threshold curves. This paper outlines an initial effort to devise such a model and to estimate how seriously noise of different kinds will limit communication.

4:15

4pAB13. Sonic boom wave propagation from air into water: Implications for marine mammals. Victor W. Sparrow (Graduate Prog. in Acoust., Penn State Univ., 157 Hammond Bldg., University Park, PA 16802)

At the 1993 NASA High Speed Research Program Sonic Boom Workshop, hosted by NASA Ames Research Center, it was noted that the development of a high speed civil transport in the next several years will focus only on overwater travel. At that meeting it was also made clear that some have raised concerns about the effects of sonic booms on the activities of marine mammals. Once a fleet of supersonic passenger aircraft are in service, there could be hundreds of flights per day, and, therefore, the effects of sonic booms on marine mammals should be well understood. The purpose of the present paper is to briefly describe the acoustic pressure and intensity variations that one can predict underwater due to a sonic boom impinging on an air-water interface. Arbitrary angles of incidence are investigated, and the proportion of energy incident and reflected from the interface are given for typical sonic boom wave forms.

THURSDAY AFTERNOON, 7 OCTOBER 1993

DENVER ROOM, 1:00 TO 4:45 P.M.

Session 4pEA

Engineering Acoustics and Structural Acoustics and Vibration: Vibration and Acoustic Monitoring of Machinery and Structures

Robert D. Finch, Cochair

Department of Mechanical Engineering, University of Houston, Houston, Texas 77204-4792

Aynur Unal, Cochair

4255 Suzanne Drive, Palo Alto, California 94306

Chair's Introduction—1:00

Invited Papers

1:05

4pEA1. Using sound and vibration to diagnose machinery faults. Richard H. Lyon (MIT, Cambridge, MA and RH Corp., Cambridge, MA 02138)

The relation between the dynamical forces and structural elements of a machine and the vibratory or acoustical signals produced as it operates is far from straightforward. Nevertheless, a number of developments in measurement technology, system modeling, and signal processing have made it more possible in the last few years to extract information about both mechanism performance and structural integrity from vibration or acoustical measurements. Measurement advances include smaller and more sensitive transducers. System modeling advances include hybrid FEA and dynamic simulation methods. And signal processing includes a number of methods for amplitude and frequency demodulation and deconvolution procedures. The paper will outline a number of these advances, illustrated by some of the author's experiences.

1:30

4pEA2. Acousto-ultrasonic nondestructive evaluation of materials and structures. Henrique Reis (Dept. of General Eng., Univ. of Illinois, 117 Transportation Bldg., 104 S. Mathews, Urbana, IL 61801)

A review of the nature and the underlying rationale of the acousto-ultrasonic methodology is presented. The term acousto-ultrasonics denotes a nondestructive evaluation technique that combines some aspects of acoustic emission methodology with ultrasonic simulation of stress waves. Unlike most nondestructive evaluation techniques, acousto-ultrasonics is less concerned with flaw detection than with the assessment of the collective effects of the various flaws and materials anomalies. Acousto-ultrasonics is primarily concerned with material properties variations, such as the significant reduction in strength and toughness caused by combinations of minor flaws and diffuse flaws populations. Porosity content, and fatigue and impact damage are typical examples of factors that affect material properties variations. Applications of the acousto-ultrasonic approach to the nondestructive evaluation of wire rope, porosity in laminated composites, fatigue damage in ceramic matrix composites, and adhesive bond strength in finger joint connections of structural lumber and in other structural systems are presented and discussed.

1:55

4pEA3. Vibration and acoustic monitoring of rotating shafts. Andrew D. Dimarogonas (Dept. of Mech. Eng., Washington Univ., Box 1185, St. Louis, MO 63130)

Vibration and acoustic monitoring of the operating conditions and incipient failures of rotating shafts and other machine members has been used extensively in the past 30 years. The shaft vibration spectrum was used to obtain information about unbalance, bearing, and other instabilities in rotating machinery since the 1920's. It was further identified in the 1960's as a source

of information for the existence of cracks, misaligned couplings, loose parts, and other unwanted operating conditions. Such conditions would mainly introduce asymmetries of the rotating parts which would result in vibration spectra in the acoustic range which are characterized by the response of parametrically excited systems, including sub- and superharmonics. Such signatures have been used for rotating machinery condition monitoring. It was further observed that field singularities, such as cracks, couple the lateral, longitudinal, and torsional vibration of rotating shafts providing additional monitoring tools. Finally, it was found that the nonlinearities associated with closing cracks and gaps and journal misalignments lead to additional information on the condition of the rotating parts, such as higher harmonics. In recent years, the wealth of accumulated observation on the signatures of abnormal operating conditions in rotating machinery have been incorporated in expert systems for rotating machinery monitoring. Such systems have been developed using predicate logic or artificial neural networks. Further, fuzzy sets and interval analysis have been incorporated to account for nontraditional recording of operating conditions.

Contributed Papers

2:20

4pEA4. Decay of resonances of reinforced concrete beams with cracks. Zhijing Wang, Shail Pandya, Robert D. Finch (Dept. of Mech. Eng., Univ. of Houston, 4800 Calhoun Rd., Houston, TX 77204-4792), and Ben H. Jansen (Univ. of Houston, Houston, TX 77204-4793)

Reinforced concrete beams in practical use always contain cracks. This paper reports a study of the changes in decay rates of resonance frequencies when cracks occur in a vibrating concrete beam. The experiments were performed using model reinforced concrete beams with general end conditions as well as ideal pinned-pinned end conditions. The power spectrum and its time variation were studied. The short time Fourier transform (STFT) technique was used. Theoretical studies were made to establish the effects of damping on the vibration of the beam. A complication exists in that the reinforced concrete beam is in effect a composite material. The results will be compared with the data obtained from a steel beam of homogeneous material [J. Robin, *Acoust. Soc. Am.* **92**, 2441 (A) (1992)]. The changed behavior of decay rates of resonance frequencies could be a feature in the vibration monitoring of reinforced concrete structures such as highway bridges. [Work supported by NSF Grant No. MSS-9024224.]

2:35

4pEA5. Response of steel beams with elastically restrained end conditions with and without cracks. Shail R. Pandya, Zhijing Wang, Robert D. Finch (Dept. of Mech. Eng., Univ. of Houston, 4800 Calhoun Rd., Houston, TX 77204-4792), and Ben H. Jansen (Univ. of Houston, Houston, TX 77204-4793)

The vibration response of uniform and cracked beams with elastically restrained end conditions under impact excitation is being experimentally investigated. Preliminary work has revealed that the response obtained does not match that predicted theoretically for pinned-pinned end conditions. In practice, most beams have nonclassical end conditions as a certain amount of flexibility is unavoidable. Therefore a theory of vibration of beams with elastically restrained end conditions was applied. The results of this theory were in close agreement with the response obtained from beams mounted on elastic supports. An antiresonance for the pinned-pinned beam has been found for the fifth mode of vibration whose frequency matches the natural frequency of the supports. Cracks simulated by slots are currently being made at various locations along the length of the beam. The impact response of these slotted beams will also be reported and comparisons will be made between the theoretical and the experimental results. [Work supported by NSF Grant No. MSS-9024224.]

2:50-3:00 Break

3:00

4pEA6. Noise attenuation provided by flexible shields for industrial equipment. Natalia V. Sizova (Cooperative Inst. for Res. in Environmental Sci., Campus Box 216, Univ. of Colorado, Boulder, CO 80309-0216)

A general physical model and method for the calculation of the noise reduction provided by a flexible shield consisting of thin, rigid, narrow plates, joined together lengthwise into a chain is represented. The model selected is one plane containing a set of linear rectangular plates placed side-by-side, separated by narrow spaces. All plates are making flexural oscillation under the influence of evenly distributed noise pressure. The model is based on the determination of the sound pressure caused by sound radiation of a thin rectangular rigid plate in average frequency range (63-8000 Hz), in far field. The loss factor of the plates junction was determined experimentally and used for the calculation of the noise reduction. Some experimental data for the noise reduction provided by different designs (different number of the plates, material, thickness) were represented and compared to the calculated results. The method allows to determine the optimal parameters which are necessary to achieve maxima of the noise reduction. [Work supported by Scientific-Research, Institute of Printing Machinery, Moscow.]

3:15

4pEA7. Diagnosis of the Bjoerk-Shiley heart valve failure from acoustic signatures. Thomas Chondros (Dept. of Mech. Eng., Univ. of Patras, Patras, Greece)

The replacement of human heart valves by mechanical ones was pioneered in the 1960s. Shiley, Inc., of Irvine, California, was one of the early manufacturers of such valves. The Bjoerk-Shiley 60 convex-concave valve was introduced in 1967 and about 85 000 people received this valve until 1986, when it was withdrawn from the market. Analysis of sound emitted from the strut of the Bjoerk-Shiley 60c CC valve due to impact was used to monitor the propagation of the fatigue crack before it would lead to the failure of one or both legs of the outlet strut. Analytical and experimental results established that the range of the fundamental natural frequency is 2500 to 8000 Hz. It was found that previous experimental investigations were performed within a window of lower frequencies and they could not capture the transition of the natural frequency during the propagation of the fatigue crack, which would take several months. It was found that for crack depths above 50% of the diameter, the change in the natural frequency, thus the dominant component of the signal, was substantial and easily measurable. This may lead not only to diagnosis of the fracture at one end of the outlet strut but also to detection of the propagating crack long before it would lead to the partial or complete strut failure.

3:30

4pEA8. Simultaneous measurement of structural vibration and temperature using an embedded optical fiber in a flat plate composite structure. Bornain Chiu and Mardi C. Hastings (Dept. of Mech. Eng., Ohio State Univ., 206 West 18th Ave., Columbus, OH 43210)

Numerical results from this study show that a single polarization-maintaining optical fiber embedded in a composite flat plate is able to simultaneously measure low-frequency structural vibration and temperature. Theoretical estimates of sensitivities for the optical fiber sensor are presented with consideration of the cross sensitivity caused by simultaneous disturbances. An analysis of the composite plate and fiber

based on the theories of linear elasticity and photoelasticity is used to estimate sensitivities of the embedded fiber. A finite element analysis using ANSYS is used to determine the state of stress in the composite plate. Results of this study show that simultaneous changes in structural vibration and temperature can easily be resolved.

3:45

4pEA9. Acoustic diagnosis of the onset of strut fracture of the BS/CC prosthetic heart valve. Allen C. Eberhardt and Charles E. Chassaing (Structural Acoust., Inc., 5801 Lease Ln., Raleigh, NC 27613)

A replacement prosthetic heart valve functions in a corrosive environment in which it cannot readily be inspected or repaired. It must cycle repeatedly, at a rate of 40 million cycles per year, over a service life that may extend many years. The opening and closing sounds of Bjoerk-Shiley convexo-concave prosthetic heart valves are brief transients that are being studied for detection of incipient valve failure. The sounds are produced by the impact of the occluder disk with the metallic struts that retain it within an orifice ring. These sounds have been recorded *in vivo* for two distinct conditions or classes, valves that are intact, and valves that exhibit an in-service failure of one of two legs of a strut which is critical to the function of the valve. The small size and rapid motion of the valve occluder disk produce a broadband acoustic transient that exhibits subtle changes as a valve strut proceeds toward failure. Features are detected, and a nonlinear expansion is used to develop a set of coefficients from the time and frequency domains to separate the two conditions. The recorded *in vivo* data are used as the training set. The coefficients are then applied to *in vivo* recordings as a method of detection of a progressive failure of the valve. Results of the expansion and of a parallel approach using neural nets are shown to be most effective for patients with normal heart rhythm.

4:00

4pEA10. Effect of microphone placement on adaptive noise cancellation methods with machine ball bearing measurements to detect bearing flaws in reverberant environments. Matthew S. Carney and J. Adin Mann, III (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50010)

Sound measurements on operating machinery in reverberant environments are difficult to analyze. Adaptive noise algorithms [C. C. Tan, IECON, 703-708 (1985)] are available to reduce the amount of background noise in the measured data. Due to the reverberation of the environment, proper placement of the microphones is important to achieve the greatest noise cancellation from the algorithm. Measurements were taken on bearings with known flaws under different loads and speeds. Results of these measurements will show the effectiveness of sound pressure measurements, and sound intensity measurements, to

detect bearing flaws with the help of the adaptive noise cancellation algorithm when the microphones are properly placed.

4:15

4pEA11. The three-dimensional laser vibrometer. Joseph Vignola (SFA, Landover, MD 20785) and Brian H. Houston (Naval Res. Lab., Washington, DC 20375-5000)

In the field of structural acoustics, two of the central tools necessary for studying the physics of interactions and vibration transmission paths are a measure of structural intensity and the generation of k - ω dispersion relationships. It is particularly important in the study of fluid-loaded structures to be able to characterize both the in-plane and out-of-plane motions in order to identify mechanism conversions at apertures and discontinuities. A noncontact underwater probe for measuring simultaneously the three components of vibration induced displacement on the surface of submerged structures has been constructed. Optical fiber is used so that a compact optical probe with no electronics is the only device in the water. The probe is used to measure the three components of motion of a point on the surface of a target from a distance of approximately 2 ft. Data for acoustic studies are obtained by collecting data at a series of points on a target surface. Experimental results will be presented. An analysis of the sensitivity, frequency response, and detection threshold of the system will also be included.

4:30

4pEA12. Sensitivity of axial vibration reactions to design asymmetries in a Stirling cycle cryocooler dual-opposed compressor assembly. Fred W. Hausle (Phys. Analysis Dept., Mail Stop WE-4, Ball Aerosp. Syst. Group, P.O. Box 1062, Boulder, CO 80306)

Stringent constraints are placed upon allowable vibration reactions from cryocoolers used for space-borne optical sensors. Stirling cycle cryocoolers typically use dual-opposed compressors to virtually cancel the net axial reaction produced by the steady-state driven oscillations of paired pistons by virtue of symmetry. Under steady-state operation, harmonics of the drive frequency occur due to the nonlinear characteristics of both the working gas and the mechanical piston support spring stiffnesses. Asymmetries in the system design due to fabrication tolerances may result in intolerable net reactions primarily at the harmonics of the drive frequency. The axial dynamics of a simple two-degree-of-freedom compressor model are examined using a finite-difference time-step Runge-Kutta algorithm. The sensitivity of the axial reaction harmonics to asymmetric characteristics between the paired compressors such as gas volume, support spring stiffness and piston mass is examined. A Fourier decomposition of the limit cycle is conducted to observe the frequency content of the vibration reactions. Comparison is made with results of an application of perturbation methods. Cryocooler design parameter asymmetry tolerances based upon specified limits on peak vibration reaction responses are obtained.

Session 4pED

Education in Acoustics: Teaching Laboratory Experiments

Uwe J. Hansen, Cochair

Physics Department, Indiana State University, Terre Haute, Indiana 47809

Thomas D. Rossing, Cochair

*Physics Department, Northern Illinois University, DeKalb, Illinois 60115**Invited Papers*

1:00

4pED1. Acoustics laboratory experiments for all levels. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Acoustics is a subject that is best learned by incorporating hands-on experience in the laboratory; this applies to students majoring in music as well as in physics or engineering. The author's laboratory manual, *Acoustics Laboratory Experiments*, describes 52 experiments at several levels of sophistication. Some are intended for students with no previous experience in a physics laboratory, some for advanced undergraduates, and some for graduate students beginning their study of acoustics. Some experiments can be performed on more than one level. Several experiments will be described and demonstrated.

1:30

4pED2. Introductory acoustics laboratory experiments. Uwe J. Hansen (Phys. Dept., Indiana State Univ., Terre Haute, IN 47809)

A course on musical acoustics designed for music majors with little or no background in science serves a number of purposes. Among these is the intent to give insight into the structure of musical sound and the physical nature of various musical instruments, and to enhance the scientific literacy of music students. Meeting these objectives is enhanced significantly if students have the opportunity to learn with their hands and ears. A series of four introductory experiments will be described and illustrated. These are (1) measuring frequency with an oscilloscope, (2) complex wave synthesis with a simple keyboard, (3) the spectrum of an elastic string, (4) the inharmonic spectrum of a stiff string.

1:50

4pED3. Laboratory experiments in the acoustics course at the U. S. Naval Academy. S. A. Elder and M. S. Korman (Phys. Dept., U. S. Naval Acad., Annapolis, MD 21402)

For about 25 years the Physics Department has offered an acoustics course as a senior elective for physics and electrical engineering majors, based on the text by Kinsler, Frey, Coppens, and Sanders. A 2-h laboratory is provided each week. The aim of the lab is to illustrate the theory by teaching basic experimental measurement techniques. The specialties include a modal analysis system (B&K) to study, for example, the vibration of guitars, Pyrex flasks, and student-made "musical" triangular steel plates. Students study the directivity of speaker arrays, musical instruments, and psychoacoustic effects in a 200-Hz–20-kHz anechoic chamber. IBM PC and Mac II work stations provide instruction in hardware interfacing, signal processing, spreadsheets (for data analysis), as well as specialized techniques of FFTs and digital filtering. Work stations have a midi-interface for sound and music synthesis. Ultrasonics and underwater acoustics experiments often involve electroacoustics. Projects include measuring the effects of interference, diffraction, reverberation, and classification of material properties. In addition, each student participates in a major project, usually in a small group. Recently the class performed an acoustical evaluation of the new Bob Hope Theater of the Performing Arts on the USNA campus.

2:10

4pED4. Undergraduate experiments for duct acoustics and wind instruments. Peter L. Hoekje (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50614-0150)

The acoustics of ducts presents a rich spectrum of practical student laboratory activities, from exposition of basic wave physics to exploration of important characteristics of musical wind instruments. A variety of duct experiments will be described, extracted from a musical acoustics course for music majors and a junior level vibrations course for physics majors at the University of Northern Iowa, as well as from an advanced acoustics seminar at Northern Illinois University. Basic physical phenomena explored include resonance, wave speed, and wavelength. These can be extended to open-end corrections, pressure, and velocity in a standing wave, acoustic impedance, and the dependence of sound speed on temperature, pressure, and gas composition. Wind instrument frequency responses can be easily observed. Small bore perturbations shift normal mode frequencies, demonstrating that resonance frequencies are affected by bore shape. Many of the experiments described use an inexpensive impedance head built by students from a piezoelectric driver and an electret microphone. This device can be customized for specific apparatus, and stimulates discussion of transducer design and use. Overall, students use simple apparatus to build a foundation of conceptual understanding which can be extended to fruitful research.

4pED5. A simple demonstration of impedance matching in driven resonant tubes. Thomas B. Gabrielson (NAWC Aircraft Div., Code 5044, Warminster, PA 18974) and David L. Gardner (CDR, NOAA Pacific Marine Environmental Laboratory, Seattle, WA 98115)

Resonant tubes are often used in teaching laboratory experiments to demonstrate a wide range of topics from standing waves to lumped-parameter approximations and electroacoustic analogues. The popularity is understandable given the ease of construction and the broad applicability outside the laboratory, ranging from impedance tubes to musical instruments. While the measurement of standing waves and modes is common, the relationships, in an open-ended tube, between the tube, the radiation impedance, and the driver impedance is less frequently addressed. In this experiment using inexpensive components and a Smith chart, the resonances of a small loudspeaker and of an open tube are measured. The radiation impedance at the tube end is transformed to the driver location and the resulting reactance is compared to the loudspeaker reactance. The tube resonances are thus demonstrated to occur when the two reactances are equal, but of opposite sign.

2:50

4pED6. The acoustical waveguide. Alan B. Coppens, Anthony A. Atchley, and Eric W. Moore (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

Fundamental concepts of normal mode propagation in bounded media can be studied simply and practically in a water-filled rectangular waveguide with pressure-release boundaries. Use of broadband source and receiver allows generation and study of well-formed gated sine-wave pulses. The source is designed to excite the lowest (1,1) normal mode of the system and suppress excitation of the next few higher modes. This provides a useful frequency range for studying the behavior of the lowest mode both above and below its cutoff frequency. The cutoff frequency can be estimated by a number of techniques; this relates cutoff to several different phenomena associated with the propagation. Tone burst operation allows study of the rise time of the propagating tone burst and thereby the group speed of the carrier frequency. With an appropriate absorber at the far end of the waveguide, cw transmissions are almost purely traveling waves so that the phase speed of the input frequency can be determined. The apparatus is simple to construct and the electronics are not exotic. This experiment is a direct descendant of one that originated in the graduate lab of the physics department at Brown University approximately 35 years ago.

Contributed Papers

3:10

4pED7. Linearized least-squares data analysis. S. L. Garrett (Phys. Dept.—Code PH/Gx, Naval Postgraduate School, Monterey, CA 93943)

One of the most attractive features of many acoustic teaching laboratory experiments is their ability to produce high-quality (precision) data. This affords the possibility of also using the results of these laboratory exercises to teach advanced techniques for data analysis that exploit the ubiquity of least-squares data analysis routines which are now included in most hand-held calculators and all personal computer plotting packages. This talk will concentrate on the transformation of general two-parameter nonlinear equations into linear forms suitable for least-squares-fitting techniques. This transformation technique will be demonstrated to (i) extract the effective moving mass of the spring to correct the simple $\omega = (k/m)^{1/2}$ expression for the simple harmonic oscillator; (ii) extract the distance from the end of the pendulum string to the pendulum bob center-of-mass in the simple pendulum experiment; (iii) extract the location of the acoustic center of a projector from measurement of the free-field pressure versus separation; and (iv) extract the cutoff frequency and thermodynamic sound speed from the phase speed versus frequency measurement in a water-filled, pressure-released waveguide.

3:25

4pED8. A simple water-filled compliant-tube calibrator. David L. Gardner (CDR, NOAA Pacific Marine Environmental Lab., 7600 Sand Point Way, N.E., Seattle, WA 98115) and Thomas B. Gabrielson (NAWC Aircraft Div., Warminster, PA 18974)

Several geometries for low-frequency underwater calibrations are in common use, but often require either anechoic walls, rigid walls, or tanks (or lakes) of large dimensions with respect to a wavelength. Such facilities are frequently not practical for use in teaching laboratories either because of inaccessibility or because the facility must be shared with active researchers. The compliant, water-filled cylindrical tube described in this paper permits low-frequency calibrations in a laboratory

environment, with modest cost and size. The sound speed inside one such device is approximately 300 m/s, which allows 100-Hz measurements in tubes 1 m long. The limitations of the compliant tube calibrator and examples of measurements made using the calibrator will be discussed.

3:40

4pED9. Continuing development of the acoustics laboratory at the Cooper Union. Daniel R. Raichel (Albert Nerken School of Eng., The Cooper Union for the Advancement of Sci. and Art, 51 Astor Pl., New York, NY 10003)

As a result of the addition of new equipment, the scope of the acoustics laboratory curriculum at the Cooper Union is being widened to include the use of FFT analyzers, reverberation meter, vibrating table, etc. Research projects are required of undergraduate seniors and graduate students must meet thesis requirements. The laboratory also serves as a research center for those interested in physical acoustics, environmental sciences, architectural acoustics, loudspeaker design, design of musical instruments, acoustics for medical diagnoses and treatment, and other acoustics-related topics. Newly established liaisons between the Cooper Union Research Foundation and other institutions (Riverside Research Institute, New York University Medical Center, U.S. Navy) open up additional opportunities for thesis work and provide access to even more equipment outside the laboratory. Even freshmen taking an integrated design course obtain acoustical laboratory experience by working in the anechoic and reverberation chambers in studying sound effects around an abstract sculpture being planned for installation in New Mexico. [Work supported by NSF and the New York State Science and Technology Foundation.]

3:55

4pED10. A one-year acoustics laboratory course at UCSD. Duncan McGehee, John A. Hildebrand, and Victor C. Anderson (Marine Phys. Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0205)

The University of California, San Diego offers a one-year acoustics laboratory course to both undergraduate and graduate students. The lab is structured to parallel a graduate level theoretical acoustics class taught jointly by the electrical and computer engineering department, and the Scripps Institution of Oceanography. The acoustics laboratory course examines fundamental principles of acoustics with emphasis on oceanographic applications. Also stressed are methods for handling random data and the professional reporting of results. A GPIB bus computer is used to control laboratory instrumentation, and students develop application programs for each experiment. The course is divided into 17 separate experiments. The first quarter is devoted to the measurement of mechanical impedance and resonance in a spring-mass system, velocity and dispersion in strings and bars, and normal modes in a bar. The second quarter examines the calibration of transducers, including self-reciprocity and beam-pattern measurements, reflection and transmission in multiple media, sound-speed dependence on salinity, and normal modes in a channel. The third quarter emphasizes the treatment of random data including: Power measurements of band-limited white noise, scattering of sound from surfaces of varying roughness, and detection threshold experiments.

4:10

4pED11. Laboratory experiments on the interaction of an air jet with a Helmholtz resonator. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402)

The interaction of an air jet with a resonator is fundamental to the operation of several musical instruments and a significant topic in musical acoustics courses. Recent studies of a Helmholtz resonator excited by an air jet have explored the dependence of the frequency and amplitude of the Helmholtz mode on jet speed and other parameters. [R. Khosropour and P. Millet, *J. Acoust. Soc. Am.* **88**, 1211-1221 (1990); J. P. Cottingham *et al.*, *J. Acoust. Soc. Am.* **92**, 2380 (1992)]. It has been observed that domains of jet speed, for which a single-frequency Helmholtz mode occurs, are separated by narrow transition regions. Investigation of this phenomenon, which is audible as well as measurable, has been adapted as a set of laboratory exercises for use in musical acoustics courses. Using a homemade air jet with the laboratory compressed air supply, students are able to study the internal sound field of a jet-excited resonator with a small tie clip microphone. Investigation of a significant nonlinear phenomenon is thus possible with simple equipment.

4:25

4pED12. MultiSensory Sound Lab for Educational and therapeutic applications. Norman Lederman (Oval Window Audio, 33

Wildflower Ct., Nederland, CO 80466) and Kimberly V. Fisher (Univ. of Oklahoma, Oklahoma City, OK 73190)

The MultiSensory Sound Lab electronically processes sound signals from microphones, musical instruments/recordings, electronic stethoscopes, and other sources and directs them to loudspeakers, a vibrating floor, and various large visual technologies that display the frequency, spectral, intensity, and time aspects of the signal. Originally developed for use with deaf children, the lab provides visual and tactile information about sound that is useful in a wide range of applications ranging from instruction in acoustics and physics of sound, to speech and music therapy for normally hearing, as well as, hearing-impaired students. Example clinical and educational applications will be presented. Data from pre- and post-testing show that sound lab activities result in significant concept learning and skill acquisition. User surveys shows that the sound lab eases teaching tasks and motivates students. Note: this presentation will serve as introduction to the MultiSensory Sound Lab tours that will be conducted at the University of Colorado. [Project supported by the U.S. Department of Education.]

4:40

4pED13. Vibration analysis of an alpine ski. Richard Chapman, Karl-Arne Hegewald, Steve Humphreys, and Scott Meckstroth (Dept. of Mech. Eng., Univ. of Colorado, Boulder, CO 80309)

An investigation into the vibrational behavior of an alpine ski excited by a single-point force applied at the tip of the ski was conducted. The single-point force was used to model small amplitude perturbations in the snow for a flat running ski to investigate ski chatter. In this respect, the bindings were assumed to remain in constant contact with the snow surface and the test ski was clamped at these locations for the experiment. An accelerometer and a personal computer were used to measure the ski's response to tip excitation and a FFT was used to determine the ski's harmonics. Numerical models were constructed utilizing (i) a finite element approximation, and (ii) a finite difference solution, to determine frequencies at 21 distinct points of the ski. Results from the model were then compared with experimental data and the model was validated. Experimental results showed the first three harmonics for the front of the ski to be 2 ± 0.5 Hz, 22 ± 1 Hz, and 43 ± 4 Hz. This compared with theoretical values of 5, 25, and 37 Hz. Experimental results for the higher harmonics and other parts of the ski were found to be within an order of magnitude of the theoretically predicted values.

Session 4pPA

Physical Acoustics: Physical Acoustics for Materials Characterization II

Dale E. Chimenti, Chair

*Department of Aerospace Engineering and Engineering Mechanics, Iowa State University, Ames, Iowa 50011**Invited Papers*

1:30

4pPA1. What physical acoustics can tell us about fluids. Robert T. Beyer (Dept. of Phys., Brown Univ., Providence, RI 02912)

Traditionally, one has used physical data on fluids to study the propagation of sound in them. However, the opposite is also possible: To use the study of the propagation of sound in a fluid to determine some of its physical properties and its behavior. This was perhaps done for the first time by Einstein (for propagation in gases) in 1920. Today, physical acoustics can be used to gain information about phase transitions in liquid crystals, to study the behavior of mixtures of liquids, and of particles or bubbles in liquids, and in other ways. A review is given of these processes and the ways in which physical acoustics contributes to our knowledge of them.

2:00

4pPA2. Thermophysical properties of alternative refrigerants from acoustical measurements. Michael R. Moldover (Thermophys. Div., Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899)

Acoustic resonance techniques have been developed at NIST to make accurate ($\pm 0.01\%$) speed-of-sound measurements in gases a routine matter. These techniques have been applied to many environmentally acceptable candidate replacement refrigerants. The data were used to deduce ideal-gas heat capacities and virial coefficients. These results are an important component of the computer package REFPROP that is being used by designers of refrigeration equipment. NIST is actively involved in developing acoustic methods for determining the viscosity and thermal conductivity of gases. NIST is also extending these measurement techniques to corrosive gases and to gases at high temperatures.

2:30

4pPA3. Ultrasonic dispersion resulting from inelastic gaseous collisions and some inferences regarding molecular interactions and force laws. Robert C. Amme (Dept. of Phys., Univ. of Denver, Denver, CO 80208)

Ultrasonics has been, and remains, a valuable technique for the investigation of vibrational energy transfer in polyatomic gases. The celebrated theory of Schwartz, Slawsky, and Herzfeld (SSH) has proven to be greatly successful in predicting the efficiency of V-T processes, particularly for self-collisions of diatomic species or between these species and the noble gases. For more complex molecules, particularly those containing hydrogen atoms, the role of vibrational amplitudes and of rotational (R-T and V-R,T) energy transfer effects may complicate the picture. Moreover, the presence of multiple relaxation processes may cause further departure from single relaxation steps in the sound velocity measurements. Nevertheless, the basic approach of SSH theory remains qualitatively correct in many cases, once the proper relaxation process and the interaction potential are identified. Vibrational collisions numbers Z_{10} , in polyatomic molecules, i.e., the average number of collisions required for the vibrational transition $v=1 \rightarrow v=0$, will be discussed both for room temperature ultrasonic measurements and, for some species, over a range of temperatures. Various predictive schemes will be reviewed, and some recent results on substituted ethanes and their mixtures will be presented. It will be shown that the SSH formulation gives agreement with these experiments when a very steep repulsive potential is employed.

3:00-3:15 Break

Contributed Papers

3:15

4pPA4. Ultrasonic nondestructive evaluation of thin (subwavelength) coatings. C. Zhu and V. K. Kinra (Ctr. for Mech. of Composites, Dept. of Aerosp. Eng., Texas A&M Univ., College Station, TX 77843-3141)

This paper describes a technique for ultrasonic nondestructive evaluation of a thin coating (subwavelength) on a thick substrate. A plane longitudinal wave which is normally incident upon the coating is considered. Transfer functions have been derived for both the coating-side

and the substrate-sideinsonification. A systematic analysis of the sensitivity of the transfer functions to the thickness and wavespeed has been carried out. An inverse algorithm has been developed to reconstruct the thickness and the phase velocity through a comparison of the theoretical and the measured transfer functions. Using this technique both the thickness and the wave speed of the coating can be extracted from the same measurement, without knowing either. The technique was used to measure the thickness and wave speed of epoxy and Plexiglas coatings (50-100 μm) on an aluminum substrate using low-frequency (10 and 20 MHz) transducers; the ratio of thickness/wavelength was about $\frac{1}{3}$. The precision in the measurement of the thickness and the wave speed

was found to be $\pm 2 \mu\text{m}$ and $\pm 3\%$, respectively. [Work supported by Texas Advanced Technology Program.]

3:30

4pPA5. Attenuated leaky Rayleigh waves. Quan Qi (Dept. of Theor. and Appl. Mech., Univ. of Illinois, Urbana, IL 61801)

The attenuation of leaky Rayleigh waves due to viscous damping in a boundary layer at the interface of an elastic solid half-space and a fluid half-space is studied by matched asymptotic method. Viscosity of the fluid is considered unimportant except in a thin boundary layer at the interface. By keeping the leading order effect, shown by the inverse of square root of a Reynolds number based on the shear velocity of the solid substrate, a new characteristic equation is obtained. One of the numerically obtained solutions gives the leaky Rayleigh wave speed and the attenuation coefficient due to both radiation into fluid and viscous damping of the boundary layer. It is shown that the viscous attenuation in the boundary layer may be as important as that due to radiation into the fluid half-space for some fluid-substrate combinations. These results may be used to improve our interpretation of experimental results of acoustic signature of materials. Extension of the analysis to a fluid layer instead of a half-space will be discussed. [Work supported by the Hunt Fellowship.]

3:45

4pPA6. Reduction of backscatter coefficient from pulse-echo measurements: A new formulation. Xucai Chen, Jack G. Mottley, Karl Q. Schwarz, and Kevin J. Parker (Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

A new formulation for the reduction of the absolute backscatter coefficient from pulse-echo measurements is presented. Using this formulation, the diffraction correction is straightforward. This correction is a weak function of frequency, rather than the frequency squared dependence proposed by earlier investigators. A much simpler calibration method is proposed. Initial experimental evidence using a focused transducer and polystyrene spheres as scatters is presented. The proposed formulation and polystyrene spheres are sufficiently simple as to be utilized in many laboratories, enabling cross checking and consistency between sites. [Work supported by NIH and NSF.]

4:00

4pPA7. Acoustic coupling from a focused transducer to a flat plate and back to the transducer. Xucai Chen, Karl Q. Schwarz, and Kevin J. Parker (Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

An exact solution for the acoustic coupling function from a focused transducer to a flat plate and back to the transducer (the diffraction correction function) is provided. This function is useful for the system calibration where is pulse-echo system or transmit-receive system is used. Numerical results will be presented for the case where the reference plate is placed near the focal plane of the transducer. The solution for a flat disk transducer is obtained as a limiting case. [Work supported by NIH and NSF.]

4:15

4pPA8. A simple apparatus for measurement of the acoustical impedance of membranes, plates, and the bulk properties of porous materials. Jason McIntosh (Digisonix, Inc., 908 Stewart St., Madison WI 53713)

An apparatus has been built that can measure the acoustical impedance (Z) of a variety of acoustical objects including membranes, plates, and porous materials. It is also possible to use the technique to measure the complex bulk modulus (B) of air within a porous material. When B is combined with Z , a complete linear description of the bulk properties of porous material is obtained. The apparatus is very simple, consisting

of two volumes, one of which is driven by a loud speaker at frequencies low enough so that the two volumes can be accurately modeled as lumped elements. It is shown that Z and B primarily depend upon on the pressure transfer function between the two volumes, which can be measured very quickly and accurately with modern analyzers. Due to the simplicity of the apparatus, it is felt that this technique is superior to methods previously used [J. D. McIntosh *et al.*, *J. Acoust. Soc. Am.* **88**, 1929-1938 (1990)], however it is limited to lower frequencies (< 500 Hz with the current apparatus). Measured impedance data of membranes, plates, ports, perforated plates, and porous material is presented as well as bulk properties of open cell foam.

4:30

4pPA9. Characterizing scatterer size via measurements of multiple acoustic parameters. Peng Jiang and Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520)

A multiple parameter approach for characterizing mixtures has been investigated. It is found that one can characterize scatterers (droplets) in immiscible mixtures of water, fat (oil), and protein, in term of volume fraction and size, from measurements of the mixture's effective acoustic parameters (density, sound speed, the acoustic nonlinear parameter, and attenuation). Mixture laws [P. Jiang *et al.*, *Ultrasound Med. Biol.* **17**(8), 829-838 (1991)] were used to determine the composition of a mixture. A procedure for choosing a reliable composition predictor was proposed. Good agreement between the predicted compositions and the known values has been demonstrated. After the volume fraction of the scattering droplet was found, an effective radius of the droplets was obtained by using an appropriate attenuation law with the measured frequency-dependent attenuation. The deduced radius is close to that of the majority of droplets observed under an optical microscope. A discussion is given on how to apply the technique to characterize the size distribution of scatterers. [Work supported by NIH through grant R01-GM30419.]

4:45

4pPA10. Acoustic characterization of chemically altered cortical bone. Jerry J. Broz (Dept. of Mech. Eng. and The Ctr. for Acoust. Mech. and Mater., Univ. of Colorado, Boulder, CO 80309), Steven J. Simske, and Alan R. Greenberg (Univ. of Colorado, Boulder, CO 80309)

A novel combination of microscopic evaluation techniques is utilized to evaluate localized compositional and material properties of bovine femoral cortical bone. Cortical bone sections were selectively demineralized by timed immersion in supersaturated (ethylenedinitrilo)-tetraacetic acid. Optical microscopy of the sections indicated that a collagen layer of varying thickness surrounded a core of mineralized tissue. The treated sections were characterized with a scanning acoustic microscope (SAM) using a 50-MHz transducer. Based on the acoustic signals reflected from both the surface of the collagen layer and the mineralized tissue layer below, the sonic wave velocities and elastic stiffnesses of the mineralized tissue and collagen were 3.65 ± 0.12 km/s and 1.49 ± 0.06 km/s and 27.2 ± 2.5 GPa and 2.95 ± 0.26 GPa, respectively [Broz *et al.*, 12th South. Biomed. Conf., Tulane Univ. (1993)]. Following the acoustic evaluation, microhardness and elemental composition maps were obtained for the sections using a diamond pyramid indenter and a Jeol JXA-8600 Superprobe, respectively. The size and location of the demineralized regions as determined by the microhardness testing and wave dispersive analysis were in good agreement with the acoustic micrograph data. The combination of acoustic, chemical, and mechanical microscopic techniques provides important insights into the site-specific phasic properties of cortical bone.

5:00

4pPA11. Design of ultrasonic scattering experiments for tissue characterization. T. Douglas Mast and Robert C. Waag (Dept. of Elec. Eng., Univ. of Rochester, Rochester, NY 14627)

Measurements of differential scattering cross section as a function of angle and frequency make possible the separation of compressibility effects from density effects. From these measurements, one can obtain the power spectrum of compressibility variations, the power spectrum of density variations, and the cross spectrum of compressibility and density variations. Measurements of these spectra has potential use for tissue characterization on scales corresponding to the wavelengths employed. Here, concepts are presented that make possible optimal design of experiments to measure ultrasonic scattering as a function of angle and frequency. Conditions are given for emitter and detector beam patterns and time gates that yield measurement system-independent scattering data. A method is presented for minimization of the effects of statistical fluctuations in the scattering estimates by solution of an overdetermined set of equations. The limitations of such measurements are also treated. As an example, an experiment currently being designed is discussed; the goal of this experiment is to measure differential scattering cross section with greater spatial resolution than has been achieved to date.

5:15

4pPA12. Limitations introduced by material attenuation in pulse-echo systems employing nondiffracting x waves. Mostafa Fatemi (Elec. Eng. Dept., Amirkabir Univ., Hafez St., Tehran, Iran)

Ultrasonic nondiffracting beams are of interest for use in pulse-echo systems because they have long depth of field and small beamwidth. Recently, a new class of nondiffracting beams, termed the x waves, has been discovered and its properties in lossless media have been reported. In some pulse-echo applications, however, propagation media are dissipative. It can be shown that x waves diffract in such media, i.e., the beam spreads and its amplitude decays with distance. In order to limit beam spread within a reasonable range, attenuation of the media must be taken into account while designing the transducer and the excitation signal. Also, amplitude decay of the echoes must be compensated for at the receiver by using a proper time gain compensation curve. In this paper, conditions for designing an x wave pulse-echo system for use in dissipative media are described. Given the attenuation coefficient and maximum allowable beamwidth, proper values for transducer parameters, such as diameter, axicon angle, beam concentration parameter in lossless media, and the bandwidth of the interrogating pulse are discussed and determined. Also, variations of the beamwidth are investigated. It is shown that for wideband x waves, the beam spreads linearly with depth. Furthermore, analytical expressions for the time gain compensation curve are obtained. In contrast to the conventional exponential gain function, the gain function for the wideband x wave approaches a linear function of time.

THURSDAY AFTERNOON, 7 OCTOBER 1993

VAIL ROOM, 1:25 TO 3.45 P.M.

Session 4pPP

Psychological and Physiological Acoustics and Musical Acoustics: Temporal and Pitch Processing, Singing, and the Tritone Paradox

Diana Deutsch, Chair

Department of Psychology, University of California at San Diego, La Jolla, California 92093

Chair's Introduction—1:25

Contributed Papers

1:30

4pPP1. Relation invariance of expressive microstructure in music performance across moderate changes in tempo. Bruno H. Repp (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695)

Two pianists played Shumann's "Träumerei" three times *con espressione* at each of three musically acceptable tempi, and the performances were recorded in MIDI format. A perceptual test showed that artificial modification of the tempi of these performances within the same moderate range does not lead to a noticeable deterioration of performance quality. Subsequent analyses confirmed that the expressive timing patterns very nearly maintained relational invariance across tempo changes. Other temporal details, however, did not expand or contract in this fashion but were unaffected by tempo; they included chord asynchronies, tone overlaps, and possibly pedal timing. This indicates distinct levels of cognitive and technical control in music performance. Dynamic patterns were essentially unaffected by tempo changes, though a slight increase in overall dynamic level with tempo was noted. The generality of these findings needs to be tested with other types of music.

1:45

4pPP2. Perception of just-noticeable time displacement of a tone presented in a metrical sequence at different tempos. Anders Friberg (Dept. of Speech Commun. and Music Acoust., Royal Inst. of Technol., Box 70014, S-100 44 Stockholm, Sweden) and Johan Sundberg (Royal Inst. of Technol., Stockholm, Sweden)

The jnd for a perturbation of the timing of a tone appearing in a metrical sequence was examined in an experiment, where 30 listeners of varied musical background were asked to adjust the timing of the fourth tone in a sequence of six, such that they heard the sequence as perfectly regular. The tones were presented at a constant interonset time that was varied between 100 and 1000 ms. The average jnd was found to be about 10 ms for tones shorter than about 240-ms duration and about 5% of the duration for longer tones. Subjects' musical training did not appear to affect these values.

2:00

4pPP3. A psychological experiment on quantitative judgment for transposition of melodies. Tamotsu Shirado and Masuzo Yanagida (Kansai Advanced Res. Ctr., Commun. Res. Lab., 588-2, Iwaoka, Nisi-ku, Kobe, 651-24 Japan)

Novice musicians have difficulty in judging the interval between a melody and its transpose quantitatively. It is expected that the more one accumulates musical experiences, the more he or she can improve their ability of quantitative judgment for the interval between a melody and its transpose. It is supposed that this ability depends on abilities of absolute pitch. Psychological experiments were conducted to investigate relations among these abilities. Single tones, pairs of single tones, and pairs of melodies and their transposes generated by a synthesizer were used for the experiments. Subjects (students majoring in music) were asked to give pitch names of single tones presented in isolation, to judge

intervals between two single tones presented sequentially, and to estimate the interval between a melody and its transpose presented on a continuous rhythm pattern. From the experiments, it can be seen that there are various types of difficulties in judging musical intervals between two single tones and those between melodies and their transposes. Further, weak correlation has been found between the ability to judge intervals between two single tones, and the ability to judge intervals between a melody and its transpose.

2:15

4pPP4. The interaction of pitch, harmonics, and the equal loudness curves in timbre analysis of musical tones. James M. Pyne (School of Music and Ctr. for Cognitive Sci., Ohio State Univ., 1855 College Rd., Columbus, OH 43235)

Highly skilled musicians encounter great difficulty when attempting to communicate information concerning the timbre of musical tones. This can be a barrier to effective teaching in that musicians tend to think of individual sounds as unified single events rather than complex compounds that establish timbre characterizations. When taken into the realm of visual representation, understanding can be more easily achieved. The utilization of a Spectral Dynamics real-time analyzer has allowed coached student performers to react to a spectral display and has proven useful in tone quality development. Of special concern has been the reduction of excessive harshness when producing loud tones, and clarity when producing soft tones, both common problems in performance. Further definition concerning the interaction of pitch, harmonic content, and the equal loudness curves has been useful in a better mutual understanding of this process and it may also be pertinent for traditional music theorists as well.

2:30

4pPP5. Pitch center of musical sounds with vibrato. Judith C. Brown (Media Lab., MIT, Cambridge, MA 02139 and Wellesley College, Wellesley, MA 02181) and Kathryn Vaughn (MIT, Cambridge, MA 02139)

The determination of the pitch center of sounds that are frequency modulated has been the focus of a number of previous studies. The sources have usually been pure tones or synthetic complex sounds with a well-defined spectral composition. In a musical context these synthetic sounds differ in temporal and spectral properties from the natural sounds produced by stringed instruments; and it is these natural sounds which performers are trained to produce and to perceive in order to make intonation choices. Samples consisting of approximately 1 s of acoustic sounds produced by a virtuoso violist playing the notes D4, C5#, A5, and G6 with and without vibrato have been chosen for this study. The sounds without vibrato were then resampled to give frequencies from -15 to +21 cents, with respect to the geometric center of the sound, with vibrato. Experiments with forced choice for pairs of sounds using musically experienced listeners (including the violist who produced the original sounds) as subjects will be reported. These experiments include a control set consisting of the comparison of pitch levels of these same sounds played without vibrato. Preliminary results with listeners who are not string players are in accord with previous results, finding that the pitch perceived is that of the geometric average.

2:45-3:00 Break

3:00

4pPP6. Effects of vibrato and vowel matching on choral blend: A correlation of spectral analyses and perceptual evaluations. Lawrence R. Brown (The Recording and Res. Ctr., The Denver Ctr.

for the Performing Arts, 1245 Champa St., Denver, CO 80204)

Four singers (soprano, alto, tenor, and bass) simultaneously recorded unison /a/ vowels on middle-C (261 Hz) in seven different conditions: No instruction, matching or aggregating vibratos, matching or aggregating a nonvibrato tone quality, matching vowel qualities with and without vibrato, and prescribed instructions for a "target" vowel with and without vibrato. A perceptual panel of choral directors, singers, and naive auditors rated the blend or homogeneous quality of randomized samples of the unison vowels. Spectral analyses of the unison /a/ samples indicated varying deviations of the actual frequency of the first 14 partials (0-4000 Hz) from the calculated integer multiples of the fundamental frequency. The tabulated results of the auditors were correlated with the percentage error between actual and calculated partial frequencies. [Work supported by NIH Grant No. R08 DC01150-03.]

3:15

4pPP7. The tritone paradox and speakers' voice range: A dubious connection. Bruno H. Repp (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695)

Deutsch and her co-workers [Music Percept. 7, 371-384 (1990); 8, 335-347 (1991)] have proposed that individual differences in the perception of the so-called tritone paradox (i.e., the perceived direction of pitch change in pairs of Shepard tones six semitones apart) derive from listeners' reference to a template acquired through experience with the pitch range of their own voice. Deutsch has reported a striking difference in tritone perception between American and British listeners, as well as a correspondence with the upper limit of the voice pitch range within an American group. The present study compared groups of Dutch, British, and American listeners. Contrary to Deutsch's observations, the perceptual results of these three groups were very similar, and there was no correlation with individual differences in vocal range within any group. Instead, there were large and systematic differences as a function of stimulus characteristics (spectral envelope), which further contradicts Deutsch's findings. These results suggest that, rather than deriving from a language-based pitch template, the perception of tritone stimuli depends on psychoacoustic factors and individual differences in auditory processing whose nature is not well understood at present. [Work carried out at IPO, Eindhoven.]

3:30

4pPP8. A regional difference within the United States in perception of the tritone paradox. Frank Ragozzine and Diana Deutsch (Dept. of Psychol., Univ. of California at San Diego, La Jolla, CA 92093)

A previous study [Deutsch, J. Acoust. Soc. Am. 88, S1 (1990)] reported a striking difference in perception of the tritone paradox between subjects who had grown up in California and those who had grown up in the South of England: When the Californians tended to hear the pattern as ascending the English group tended to hear it as descending, and vice versa. One of us (FR), who had grown up in Mahoning and Trumbull Counties in Ohio hears the pattern in a fashion typical of the southern English rather than the Californians. A study was therefore undertaken to examine perception of this pattern in subjects from this region. A statistically significant difference was found between those subject whose parents had also grown up in this region and those whose parents had not. The former group formed a bimodal distribution, and approximately half the subjects producing a histogram similar to that obtained from Californians and the others producing one similar to that obtained from the southern English. In contrast, those in the latter group produced a histogram uniformly similar to that obtained from Californians. This demonstrates regional differences in perception of this pattern, and also an effect of familial background.

Session 4pSA

Structural Acoustics and Vibration: Radiation and Scattering Theory

Joel M. Garrelick, Chair

Cambridge Acoustical Associates, 200 Boston Avenue, Medford, Massachusetts 02155

Contributed Papers

1:30

4pSA1. Convolution formulation of leaky wave contributions to scattering by plates and by cylinders and shells of variable curvature. P. L. Marston (Appl. Res. Lab., Univ. of Texas, Austin, TX 78713-8029 and Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

A novel high-frequency formulation is investigated that approximates the leaky wave amplitude at the scatterer in terms of a spatial convolution of the local incident wave pressure and a one-sided line response function $h(s>0) = -j\alpha \exp(-\alpha s + ik\rho)$. Here, s is the propagation distance along the flat or curved surface, α is the reciprocal of the attenuation length, k_l the real part of the wave number, and $j=1$ for equal fluid loading on both sides of a plate but $j=2$ for one-sided fluid loading of a shell or for Rayleigh waves on a solid. Application to plane waves incident on cylindrical surfaces (empty shell or solid) of slowly varying curvature yields the following far-field amplitude from a leaky ray propagating a distance S on the surface: $p_l = -2\alpha p_{inc} (2\pi a_1 a_2 / kr)^{1/2} \exp(-\alpha S + i\pi/4 + i\eta)$, where a_1 and a_2 are the radii of curvature at the launching and detachment regions and η is a geometrical phase accumulation. When $a_1 = a_2$, the coupling coefficient G_l for a circular cylinder derived previously is recovered. The result can be modified to situations where α varies weakly with curvature. [Work supported by ARL:UTIR&D Program and by ONR.]

1:45

4pSA2. Retroflective backscattering of sound in water due to leaky waves on facets, plates, and corner truncations: Approximate theory. P. L. Marston, S. S. Dodd, and C. M. Loeffler (Appl. Res. Lab., Univ. of Texas, Austin, TX 78713-8029)

A leaky Rayleigh (or Lamb) wave is known to be launched on a flat elastic surface (or plate) in water when the surface normal lies near a cone whose symmetry axis gives the k vector of the incident sound. If the surface has a corner with edges meeting at angles of 90° , 45° , 30° , ..., the wave vector of the leaky wave is exactly reversed due to repeated reflections at the edges that form the corner. The resulting leaky radiation is backscattered toward the source and depends only weakly on the orientation of the corner. The high-frequency cross section is large since the outgoing wave front is flat. For a suitably cut, randomly oriented facet or metal block, this effect is more likely to be observed than the specular reflection since then the normal must lie on a narrow range of angles. An approximate geometric theory for the amplitude is given. The mechanism also applies to circular cylindrical shells with 90° truncations.

2:00

4pSA3. Scattering of waves from ribs on fluid-loaded plates. Zhang Wang and Andrew Norris (Dept. of Mech. & Aerosp. Eng., Rutgers Univ., P.O. Box 909, Piscataway, NJ 08855-0909)

A new formulation of the Green's function for a fluid-loaded infinite plate under line loading is presented. The Green's function is expressed explicitly in terms of a sum of the five free modes of propagation plus a

semi-infinite integral. The Green's function itself and its first and second derivatives at the loading line are obtained in analytic forms that prove to be useful in computing the scattering matrix for a rib on the plate. The scattering matrix for an isolated rib is derived. It is shown that the wave incident onto one of the n -arbitrary ribs on the plate can be written as a product of the original incident wave and a matrix, referred to as the influence matrix, which reflects the alteration of the incident wave on one rib produced by the presence of the other ribs. The influence matrix is an inverse of a matrix whose elements are the scattering matrices for single ribs and unit matrices, and it can be expanded as an infinite series which converges rapidly. Each term in the infinite series may be interpreted as a generalized ray. Numerical results are presented and compared with previous work. [Work supported by ONR.]

2:15

4pSA4. Diffraction effects from the joining of two curved plates. Douglas Rubinsky and Andrew Norris (Dept. of Mech. & Aerosp. Eng., Rutgers Univ., P.O. Box 909, Piscataway, NJ 08855-0909)

Acoustic scattering from a fluid-loaded shell consisting of two joined curved plates is considered. Asymptotic approximations are made based upon the assumption that the fluid wavelength is much shorter than the smaller radius of curvature. A plane wave in the fluid incident upon a line weld at which the shell properties and curvature are discontinuous, but the tangent is continuous, is considered. The problem simplifies by first assuming that the angle of incidence with respect to the tangent plane at the weld is not near the critical angle for supersonic membrane waves on either side of the weld. The discontinuity then generates a diffracted wave field in the fluid which is determined by solving a Wiener-Hopf problem. This background wave field in turn forces the ordinary differential equations describing the membrane waves. From their solution, diffraction coefficients are obtained for the membrane waves generated on each curved plate by the incident acoustic field. When the incident wave is close to one of the critical angles it couples directly to the shell waves via curvature, and the interaction of this launched membrane wave with the discontinuity results in scattered membrane wave fields with diffraction coefficients determined in a manner similar to the previous case. [Work supported by ONR.]

2:30

4pSA5. Traveling wave decomposition of surface displacements on a cylindrical shell: Numerical evaluation displaying guided wave properties. Gregory Kaduchak, Thomas J. Matula, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Internal displacements of spherical shells subjected to steady sound waves in water have previously been computed by other researchers [R. Hickling *et al.*, *J. Acoust. Soc. Am.* **92**, 499 (1992)]. Such displacements can be difficult to interpret with a guided wave representation because of the superposition of counterpropagating Lamb waves and the local response to the incident wave. In this research the surface displacements of a cylinder were evaluated since they can be mathematically decomposed into counterpropagating circumferential traveling

waves. (Unlike an analogous decomposition for spheres, the amplitudes in the cylinder case are regular at 0 and 180 deg.) By limiting attention to the radial displacements of the outer surface, it was also possible to display the response to short-tone bursts and thereby distinguish between the local response to the incident acoustic wave and Lamb waves launched on the shell. Both the steady-state and burst response situations clearly manifest waves launched on the shell. These can be easily seen for ka as low as 8 for a thin shell. [Work supported by ONR.]

2:45

4pSA6. Coupling and launching of supersonic waves in elastic shells. Andrew Norris and Douglas Rebinsky (Dept. of Mech. & Aerosp. Eng., Rutgers Univ., P.O. Box 909, Piscataway, NJ 08855-0909)

The interaction of an acoustic wave with a smooth thin shell in a fluid is considered. The coupling mechanism between the acoustic field and the supersonic membrane waves, both longitudinal and shear, is discussed in detail. The coupling is mediated by the shell curvature, and vanishes when the curvature vanishes. Ray methods are used to express the membrane waves by curved wave fronts with amplitudes subject to a transport equation over the curved shell surface. The coupling, and decoupling or launching, then reduces to solving an ordinary differential equation for the unknown ray amplitude. In essence, the transport equation is forced, or "beaten" by the locally phase-matched background field. Explicit expressions are obtained for the coupling and detachment coefficients on arbitrarily curved regions. These are combined, using ray theory for the propagation over the shell, to give the scattered field due to rays traveling over the shell. The general results are explicitly tested on the cylinder and sphere, for which the ensemble of surface rays can be summed into a resonance form, and numerical comparisons are made with the exact results for these canonical geometries. [Work supported by ONR.]

3:00-3:15 Break

3:15

4pSA7. Ray acoustic modeling of the surface fields of a submerged elastic cylindrical shell with hemispherical endcaps. Jin-Meng Ho (SFA, Inc., 1401 McCormick Dr., Landover, MD 20785 and Naval Res. Lab., Washington, DC 20375)

Combining the available ray acoustic algorithms descriptive of submerged thin cylindrical and spherical shells, this paper formulates the pressure and velocity distributions on a hemispherically capped cylindrical shell insonified by an obliquely incident acoustic plane wave. As the first ray acoustic model for a noncanonical shell geometry of considerable current interest, it neglects wave coupling and diffraction due to curvature discontinuities at the junctures as well as contributions from the leaky waves excited through the endcaps. This might be justifiable, because the junctions are not sharp edges (i.e., the normals are continuous) and because the cylinder under consideration is very long compared to the radius. By matching the surface wave vector along the joints, however, due account is taken of leaky wave propagation on, and radiation from, the hemispheres. The total pressure or velocity—their ray constituents—are related to each other by appropriate impedances—on the cylinder is then synthesized by a modified geometrical acoustics field, as developed recently, and a supersonic membrane wave field that accounts for the leaky surface rays successively reaching the observer along helical trajectories and great circles on the cylinder and hemispheres, respectively. Numerical implementation is also performed here. The results are compared with the reference solution generated at NRL by a boundary element code. [Work supported by ONR.]

3:30

4pSA8. Response and radiated pressure Green's functions from a fluid-loaded cylindrical shell. J. M. Cuschieri (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431) and D. Feit (David Taylor Res. Ctr., Bethesda, MD 20084)

The solution for the response and radiated pressure from a fluid-loaded, infinite, cylindrical shell with a circumferential line load, is a classical problem in structural acoustics. Although this represents a rather fundamental problem, the solution for the Green's function typically involve a contour integration for each spatial point of interest. An alternative approach has been developed which solves for the Green's functions without the need to use a contour integral, for every spatial location, and without the need to introduce structural damping. The approach is a hybrid numerical/analytical solution, where the numerical part is based on an inverse discrete Fourier transform (IDFT) and the analytical part is the contribution of some of the poles in the solution which have to be removed to eliminate the singularities from the integrand of the IDFT. The singularities in the integrand are attributed to the real and near real poles of the fluid-loaded cylindrical shell response in the wave number domain. The location of the singularities is obtained using a numerical search algorithm and therefore the developed solution can be used to more complex geometries involving the fluid-loaded cylindrical shell. Results will be presented which match well with results found in the literature. [Work sponsored by ONR.]

3:45

4pSA9. Structural acoustics of cylindrical shells having multiple layers using the direct global matrix. David C. Ricks and Henrik Schmidt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

A method has been developed to compute the response of cylindrically layered viscoelastic systems using the direct global matrix approach (DGM). The numerical stability of the method allows us to model multilayered shells that have viscoelastic layers for damping and compliant coatings for decoupling. In this presentation, the method is applied to a cylindrical shell having external fluid loading and a viscoelastic layer of damping material sandwiched between two solid layers. The model can be excited by ring forces, point forces, and a point source in the fluid. Parameter studies are presented to investigate the effect of the viscoelastic damping layer on the various wave types (flexural, membrane compression, and membrane shear). [Work supported by ONR.]

4:00

4pSA10. Forward modeling of midfrequency wave propagation in shells. Joseph E. Bondaryk and Henrik Schmidt (Dept. of Ocean Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Experimental, bistatic, scattering measurements were made on two, fluid-loaded, cylindrical shells in the frequency range of $2 < ka < 10$. Data were collected for an empty shell and a duplicate shell stiffened with unequally spaced ring stiffeners and resiliently mounted, wave-bearing, internal structural elements. The shells were insonified by wideband pulses of spatially plane waves at bow incidence and the scattered field was measured by a synthetic array in the transition field regime. Time domain focusing is used to backpropagate the measured field to the shell surface. The specular contribution to the measured field is first estimated and removed to eliminate it in the backpropagated data. The resulting waveforms show the transient, dynamic, structural response of the shell, uncluttered by the specular wave and its sidelobes. The measurement array does not provide enough resolution to allow direct analysis of membrane waves traveling on the shell. Therefore, a forward-modeling technique coupling a transmission-line model of the shell and simulated annealing parameter estimation is used. This procedure provides estimates of wave speeds, decay rates, and reflection and transmission coefficients at shell discontinuities for flexural and compressional waves on the both shells. [The authors acknowledge NRL for acquisition of scattering data. Research supported by ONR.]

4:15

4pSA11. Effects of internal attachment on sound scattering from cylindrical shells. Y. P. Guo (Dept. of Ocean Eng., Rm. 5-204, MIT, Cambridge, MA 02139)

This paper discusses the effects of attachment conditions of internal structural loading on sound scattering from cylindrical shells. The internal loading is modeled as circular elastic plates. It will be shown that strongest scattering results from attachment that constrains tangential motions in the shell, because this kind of attachment interacts with compressional and shear waves most strongly, which are dominant waves contributing to the scattered field and are essentially associated with tangential motions in the shell. This is in sharp contrast to sound scattering from structural joints in plane geometries, a flat plate, for example, where the scattered field is dominated by constraints that restrict motions normal to the plate. This suggests that sliding joints in shell like structures should be acoustically beneficial.

4:30

4pSA12. On the acoustic advantages of a helical-ring stiffener in a thin cylindrical shell. Y. F. Hwang (David Taylor Model Basin, Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

A conventional circular-ring stiffener in a thin cylindrical shell may cause undesirable sound radiation due to the scattering of subsonic flexural waves. Vibration in each of a set of disjointed circular rings may stay within the ring, which results in a resonant buildup that acts as a localized excitation of the shell. Furthermore, elastic waves in a cylindrical shell tend to propagate, having a plate-wave front, along the axis of the cylinder. A circular ring is oriented in the plane normal to the direction of wave propagation and therefore it sees the elastic wave as a normal incident wave. These rings are separated at a certain distance in tandem. An incident wave in the shell may be reflected back and forth between a pair of adjacent rings. Therefore, a circular ring may not only be an efficient scatterer, it could also scatter the same wave several times causing a reverberant buildup in the shell between rings, and, consequently, enhance acoustic radiation from the shell. This paper discussed the technical rationale to alleviate these shortcomings by using a continuous helical-ring stiffener in a thin cylindrical shell.

4:45

4pSA13. Waves in a cylindrical shell stiffened by a helical rib. K. Steven Kim (Signatures Directorate, Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

In vacuo elastic wave propagation characteristics in an infinite cylindrical shell stiffened by an infinite helical rib are investigated analytically. Equations of motion of the helical rib and of the cylindrical shell in helical coordinates are developed. They are combined into a single set of equations using appropriate boundary conditions. Dispersion curves of the elastic waves propagating in the shell are obtained for different

pitch angles. Comparisons with wave propagation characteristics in an unstiffened shell are discussed.

5:00

4pSA14. Response of submerged cylindrical shells with internals to an impulse line load. Michael J. Utschig, Takeru Igusa, and Jan D. Achenbach (Dept. of Civil Eng., Northwestern Univ., Evanston, IL 60208)

An infinitely long cylindrical shell containing internal substructures and submerged in a fluid is excited by a short duration line load. The response of the shell, substructure, and fluid is examined in the frequency and time domains. To gain insight into the response characteristics, two approaches are used. The first is the normal mode approach, where the response is expressed as a Fourier series in the circumferential coordinate [M. C. Junger and D. Feit (1986)]. In the second approach, the shell is unwrapped into a two-dimensional manifold unbounded in both the longitudinal and circumferential coordinates [Pierce and Kil, *J. Vib. Acoust.* **112**, 399-406 (1990)]. The response is expressed as a Fourier transform in the circumferential coordinate. The results of the modal approach are useful in examining resonances between the fluid-loaded shell and the substructures and in interpreting the effects of the substructure connections. The results of the second approach can be visualized as waves propagating around the shell circumference and generating near-field acoustic disturbances. A time sequence of contour plots of the near-field pressure reveals the evanescent and propagating acoustic waves associated with the shell and substructure responses. [Work supported by ONR.]

5:15

4pSA15. Echoes from a submerged spherical elastic shell coupled to an internal mechanical system. H. Huang and G. Gaunaud (Naval Surface Warfare Ctr., White Oak Detachment, Silver Spring, MD 20903-5640)

The scattering of sound waves by a submerged spherical elastic shell coupled to an internal two-degrees-of-freedom mechanical system is studied. The problem is solved using the classical separation-of-variable technique taking into account the effects of the trilateral interaction between the incident wave, the elastic shell, and its internal mechanical system. Numerical results show that echoes from the shell varies significantly with the mechanical properties of the internal system. Particularly, the analytical solution indicates that at the "fixed base" resonance frequencies of the internal system, the shell motion stops at the point where the internal system is attached. The scattered acoustic pressure fields at these frequencies are analyzed in detail. [Work supported by NSWC.]

Session 4pSP

Speech Communication: Phonetic Perception

Susan Nittrouer, Chair

Boys Town National Research Hospital, 555 North 30th Street, Omaha, Nebraska 68131

Contributed Papers

1:00

4pSP1. Trade-off between burst cues and transition cues in the perception of prevocalic stop consonants: Predictions from acoustical measurements. Roel Smits and René Collier (Inst. for Percept. Res., P.O. Box 513, 5600 MB Eindhoven, The Netherlands)

Two male Dutch talkers produced two tokens of each of 20 stop-vowel syllables (/b,d,p,t,k/ followed by /a,i,y,u/). The release bursts were separated from the voiced parts and four types of stimuli were created: Burst-only stimuli (BO), burstless stimuli (BL), stimuli with cross-spliced bursts, where burst and transitions indicate conflicting place-of-articulation information (CS), and original utterances (OR). These stimuli were presented to 20 subjects for identification. Results show the well-known vowel-dependent trade-off between burst cues and transition cues. An attempt was made to predict the confusion matrices from acoustic properties of release bursts and formant transitions using Luce's similarity-choice model. Most of the variance (89%) in the confusion matrix for the BO condition was explained using a spectral tilt measure and a spectral compactness measure. Much of the variance (80%) for the BL condition was explained using frequencies of F_2 and F_3 at voicing onset and of F_2 in the vowel. Using the same burst and transition measures for the CS condition the explained variance dropped to 40%. Preliminary results show better predictions when acoustic measures are used which integrate over burst and transitions.

1:15

4pSP2. Acoustics of perceptual centers in speech. Charles A. Harsin (Dept. of Psychol., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148)

An investigation of the acoustic correlates of perceptual centers (p centers) in CV and VC syllables was conducted. Subjects located the p centers of the syllables by placing the syllables into perceptual isochrony with a series of 1000 Hz, 5-ms clicks. In all syllables the vowel was /a/. The CV syllables used the consonants /s/, /r/, /n/, /l/, /d/, /k/, and /g/, while the VC syllables used the consonants /s/, /r/, and /n/. Consonant duration was varied within each VC combination and within the CV combinations which did not begin with a stop. Energy modulations of between 3 and 42 Hz were extracted from the syllables and were weighted according to an acoustic modulation sensitivity function. The rate, magnitude, and location of energy modulations within the syllables were all found to affect p-center location. In each syllable, the largest energy modulation, which was associated with the vowel onset, had the greatest effect on p-center location. In CV syllables, the modulation associated with the consonant had a secondary effect, while in VC syllables the initial vowel modulation was apparently the only determinant of p-center location. The nature of the relevant modulation energy parameters suggests that they could be acting as information about velocities and/or accelerations of articulators, and that p-centers might signify an integration of multiple articulatory events into a single syllabic event.

1:30

4pSP3. Perception of the [m]-[n] distinction in VC syllables produced by child and adult speakers. Ralph N. Ohde, Christine W. McMahon, and Katarina L. Haley (Div. of Hear. and Speech Sci., Box 552, Station 17, Vanderbilt Univ. School of Medicine, Nashville, TN 37232)

This research extends previous developmental studies on the perception of the [m]-[n] distinction in CV syllables [R. N. Ohde and K. Haley, *J. Acoust. Soc. Am.* **92**, 2463(A) (1992)]. With only one exception, three talkers for each age level of 3, 5, 7, adult female, and adult male produced VC syllables consisting of either /m/ or /n/ in the context of four vowels /iaəua/. Two productions of each syllable were modified using waveform editing techniques so that the distribution of place of articulation cues for consonant perception could be determined. Ten adults identified the place of articulation of the nasal from several murmur and vowel transition segments. Preliminary findings indicate that the salience of the place of articulation feature from spectral discontinuity cues is substantially weaker in VC syllables than CV syllables, particularly in children's productions. The findings for the perception of speech segments will be discussed relative to (1) The independence of formant transitions in the vowel and the murmur spectrum as cues to place of articulation in VC syllables, and (2) the developmental role of spectral change between the vowel and the murmur in syllable acquisition. [Work supported by NIH, DC00464.]

1:45

4pSP4. A new masked spectrum representation applied to English /r/-/l/ dissimilarity measurement. Kiyooki Aikawa and Reiko A. Yamada (ATR Human Information Process. Res. Labs., 2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto, 619-02 Japan)

This paper proves that a dynamic cepstrum is effective not only in automatic speech recognition, but also in explaining the speech perception mechanism. Talker dependency on the perception of American English /r/-/l/ in Japanese listeners has been reported [J. S. Logan *et al.*, *J. Acoust. Soc. Am.* **89**, 874 (1991)]. In order to interpret this phenomenon with regard to the acoustical properties of the stimuli, talker dependency is assumed to be caused by the acoustical dissimilarity of the stimuli. This paper applies a dynamic cepstrum to measuring the dissimilarity of the stimuli. The dynamic cepstrum is a new spectral representation for automatic speech recognition that incorporates the time-frequency characteristics of forward masking [K. Aikawa *et al.*, *J. Acoust. Soc. Am.* **92**, 2476(A) (1992)]. This parameter enhances formant shifts and suppresses stationary spectral features. In a perception experiment, identification tests for English /r/-/l/ minimal pairs uttered by five talkers were conducted. The dynamic cepstrum and a conventional cepstrum were compared for their performance in measuring the acoustical dissimilarity of the minimal pairs. The /r/-/l/ dissimilarity measured by the dynamic cepstrum shows a talker dependency highly correlated with the talker dependency of the perception experimental results. [Speech data were provided by Dr. D. B. Pisoni, Indiana University.]

4pSP5. The perceptual weighting of acoustic cues changes with linguistic experience. Susan Nittrouer, Carol Manning, and Gina Meyer (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131)

Previous studies found that children's judgments of syllable-initial /s/ and /ʃ/ are more related to the vocalic F2 transition and less related to the fricative-noise spectrum than are adults' judgments [e.g., Nittrouer, J. Phon. 20, 351-382 (1992)]. Such results have led to a model of speech development proposing that children's weighting of acoustic cues changes as they gain linguistic experience. The present study tested two requisites of that model, namely that the perceptual weighting of acoustic cues must be flexible and cannot simply reflect a listener's auditory sensitivities. Adults and 3-year olds participated in two tasks: Identification tasks, using synthetic fricative noises and either natural or synthetic vocalic portions; and discrimination tasks, measuring sensitivity to fricative-noise spectrum and F2 transition. Identification tasks showed the same age-related differences found in earlier studies when natural vocalic portions were used, but these age effects were reduced when synthetic vocalic portions were used. Discrimination tasks showed slightly larger difference thresholds for both the fricative noise and the F2 transition for children than for adults, but this age effect could not explain the age effect on the weighting of those cues for identification. It was concluded that the weighting of acoustic cues is flexible both for adults and for children, and that age-related differences in the weighting of the cues to /s/ and /ʃ/ are not explained by age-related differences in auditory sensitivities. [Work supported by NIH.]

2:15

4pSP6. A mapping between trainable generalized properties and the acoustic correlates of distinctive features. Mark A. Johnson (Res. Lab. of Electron., Dept. of Elec. Eng. and Comput. Sci., Rm. 36-511, Cambridge, MA 02139)

It has been suggested that the acoustic correlates of distinctive features may be most reliably measured relative to important acoustic events, or landmarks [K. N. Stevens *et al.*, Proc. Int. Conf. Spoken Language Process. 1, 499-502 (1992)]. Theoretical and empirical literature in the field of phonetics describes many of the acoustic correlates of distinctive features, but these acoustic correlates are often described in terms of formant frequencies and amplitudes, aspiration noise, and other attributes which are difficult for a speech recognizer to measure reliably. This paper reports on a preliminary attempt to compile a table of acoustic correlates of distinctive features, and to map these correlates to reliable acoustic measurements in the vicinity of an acoustic landmark. The acoustic measurements are simple energy and spectral change measurements, defined in terms of frequency and timing parameters which may be optimized for maximum feature discrimination using the technique described by Phillips and Zue [Proc. Int. Conf. Spoken Language Process. 1, 795-798 (1992)]. The table is indexed by distinctive feature, and if an acoustic correlate is likely to be salient only in a given phonological context, the context is given as a list of neighboring and simultaneous distinctive features. [Work supported by NSF.]

2:30

4pSP7. A speeded-discrimination task is used to examine processing dependencies between pitch and place of articulation. Anu Sharma, Thomas D. Carrell, and Nina Kraus (Dept. of Commun. Sci. and Disord., Northwestern Univ., Searle Bldg., 2299 N. Campus Dr., Evanston, IL 60208)

Speeded classification of stimuli varying along two independent acoustic dimensions is frequently used to study perceptual processing of dependencies in speech perception. Reaction times are compared for different presentation sequences to infer perceptual dependencies. Initially, Wood [J. Exp. Psychol. 1, 3-20 (1975)] observed a unidirectional dependency relation demonstrating that the processing of place of articulation is dependent on the earlier processing of pitch. Subsequently,

other investigators reported diverse patterns of dependencies for different stimulus parameters. However, it is not clear if these dependencies are specific only to the speeded-classification paradigm. The purpose of this experiment was to try to generalize Wood's (1975) results using a different paradigm. The processing dependencies between pitch and place of articulation using a *speeded-discrimination* paradigm were chosen for examination. The basic principle of the experiment was the same, however instead of classifying the sounds, subjects had to discriminate pairs of sounds using a two-choice ("same" or "different") discrimination task. The overall pattern of results was similar to Wood's study, i.e., there exists an asymmetric dependency between pitch and place of articulation. Furthermore, this pattern of results was observed for "same" as well as "different" reaction times. A second finding (not reported by Wood) was that the error rate data showed the same pattern of results as the reaction time data. Finally, it appears that speeded discrimination requires a greater level of selective attention and cognitive processing for the decision making process than speeded classification.

2:45-3:15 Break

3:15

4pSP8. Dialect differences in vowel perception. Alice Faber (Haskins Labs., 270 Crown St., New Haven, CT 06511), Catherine T. Best (Wesleyan Univ. and Haskins Labs.), and Marianna Di Paolo (Univ. of Utah)

Analysis based on phonemes assumes that "u," for example, in whatever context is phonologically the same, and that deviations—sometimes extreme—imposed by following approximants do not affect the category membership of the vowel. Informed by Best's perceptual assimilation model, which predicts perception of non-native speech contrasts from their relationship to native segment inventories, this study investigates the contrasting possibility that American English contrasts like *pool-pull* and *heel-hill* are perceived independently of the contrasts between /u/-/ʊ/ and /i/-/ɪ/ in other contexts. Connecticut and Utah listeners heard productions of both contrasts by speakers from both states; the contrasts are acoustically less different for Utah than for CT speakers. Listener groups completed Keyword identification, forced-choice labeling, and AXB discrimination tasks. There were systematic listener dialect differences on the keyword and AXB tasks, and both groups did better on the CT than on the UT contrasts. *Heel* and *hill* were identified with the vowels in HID and/or HEED; however *pool* and *Npull* were identified with AWED or HOED more than with FOOD or HOOD. Even when *pool* and *pull* were identified with the same keyword, one class often contained better exemplars of the keyword than the other, and discrimination was still above chance. These results suggest that *pool* and *pull*, and *heel* and *hill*, minimally distinct in Utah, may differ primarily in their distance from /ɔ/-/o/ and /ɪ/, respectively. [Work supported by NIH Grant Nos. HD-01994 and DC-00403.]

3:30

4pSP9. Production and perception of work-initial stops by Korean adults. Mi-Ran Kim (Dept. of English, Univ. of Wisconsin, Madison, WI 53706), Charles Read, Keith Kluender, and Andrew Lotto (Univ. of Wisconsin, Madison, WI 53706)

In Korean there are three classes of stops: Voiceless unaspirated [t], voiceless slightly aspirated [t^b], and voiceless heavily aspirated [t^{hh}]. For these classes of stops produced by male and female Seoul and Pusan speakers, voice-onset time (VOT), amplitude of aspiration, and f0 of the first five glottal pulses were measured. VOT was longer for slightly aspirated than for unaspirated stops, and longer still for heavily aspirated stops. Unaspirated and heavily aspirated stops were both produced with significantly higher f0 (more than 30%) than slightly aspirated stops. Gender and dialect (Seoul versus Pusan) differences in these variables were also noted. Then, six 12-step series of syllables differing in VOT (5-82 ms), fundamental frequency (100, 125, 150

Hz), and amplitude of aspiration noise (-33 , -21 dB *re*: vocalic energy) were synthesized. These stops, varying perceptually from [ta] to [t^hba], were presented to 12 native-Korean listeners. Results indicate that all three stimulus variables were perceptually significant for the Korean listeners. [Work supported by NIDCD Grant No. DC-00719 and NSF Grant No. DBS-9258482.]

3:45

4pSP10. Consonant context affects perceived similarity of North German and American English vowels. Winifred Strange, Sonja A. Trent, Janet W. Stack, Xiang Ling, and Alba I. Rodriguez (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4204 E. Fowler Ave., Tampa, FL 33620-8150)

Several current theories of L2 learning attempt to predict relative difficulties in learning to produce phonetic contrasts of a foreign language on the basis of the non-native phones' *phonetic similarity* to native phoneme categories. In this study, monolingual speakers of American English (AE) were asked to categorize (into AE vowel categories) instances of the 14 North German (NG) vowels and to judge their goodness-of-fit on a seven-point scale. Multiple tokens of each vowel were produced in /C-vowel-t/ syllables (where C = /b,d,g/) in a carrier sentence. Results indicated that both the proportion of assignments to particular AE categories and the goodness-of-fit ratings were influenced significantly by the initial consonantal context. While front rounded NG vowels were categorized more often overall as back rounded AE vowels than as front unrounded AE vowels, front unrounded responses were significantly greater for front rounded vowels in labial context than in alveolar and velar contexts. In addition, the categorization of several NG vowels indicated significant individual differences among subjects in categorization patterns. Implications of these findings for conceptualizations of native-language phonetic "prototypes" as discussed.

4:00

4pSP11. A dynamic model for the temporal properties of Swedish. Bertil Lyberg and Barbro Ekholm (Telia Res. AB, S-136 80 Haninge, Sweden)

The segment duration varies depending on a number of linguistic and nonlinguistic factors. At the word and phrase levels, the segment duration is found to vary depending on the position in the word and in the phrase. The positional effects on segment duration have been studied by several investigators and, for some languages, models have been hypothesized in order to describe the duration of speech segments in different positions in words and phrases. The greatest positional effect is the phenomenon of final lengthening that appears to be of considerable generality as a phonetic phenomenon. Most computational models for segment duration are, however, static, i.e., the durational properties are modeled at a certain speech rate. In this investigation the segment duration is studied at different speech rate and with focus assignment systematically varied along the sentence. The segment duration values are studied by means of both reiterant speech and ordinary read speech. A tentative dynamic model for the segment duration is presented.

4:15

4pSP12. Acoustic correlates of the fortis/lenis contrast in Swiss German plosives. Sean A. Fulop (Dept. of Linguistics, Univ. of Calgary, 2500 University Dr., Calgary, AB T2N 1N4, Canada)

Features of articulatory tension in plosives, such as increased closure duration, have not been shown to be perceptually salient in the time

following the plosion (with the exception of increased F_0) but are thought to influence perception by their appearance in the closure phase preceding plosion [K. Kohler, *Phonetica* 36, 332-343 (1979)]. Kohler [Phonetica 41, 150-174 (1984)] has claimed that the two plosive classes in Swiss German cannot contrast initially (where closure features are not audible), as neither voicing nor aspiration supports the contrast. This paper reports preliminary acoustic data (spectrograms) obtained from the natural production of Swiss German plosives by a native speaker. The data indicate that sonorants following fortis consonants exhibit increased high formant clarity and intensity in relation to the first spectral peak. Kohler and W. A. van Dommelen [J. Phon. 15, 365-381 (1987)] show a similar acoustic feature to be associated with global "tense voice" manner in the vowels surrounding intervocalic plosives. It is thus suggested that the observed acoustic feature is one of tension associated with the production of fortis plosives in Swiss German, and that this is available to maintain the fortis/lenis contrast initial position.

4:30

4pSP13. Intelligibility of Mandarin speakers of English: Correlation of acoustic and perceptual measures. C. Wardrip-Fruin and A. Constantinou (Dept. of Commun. Disord. and Program in Linguistics, California State Univ., Long Beach, CA 90840)

Wave duration and wave peak amplitude were measured for 10 Mandarin speakers of English and five native English speakers. Results revealed significant differences between the two groups on these acoustic variables. Discriminant function analysis resulted in 12 (80%) of 15 subjects being classified correctly in their respective language groups based on the wave duration and wave peak amplitude measures. A second experiment, with the 10 Mandarin subjects as speakers and 231 native English speakers as listeners, ranked the intelligibility of each speaker. A regression analysis revealed high correlation ($r=0.897$) between acoustic and perceptual measures; measurements of amplitude wave duration account for 89.1% of the variation in intelligibility and measurements of peaks in wave amplitude account for 68.5% of the variation in the speech samples. These findings suggest that intelligibility of Mandarin speakers of English is a function of acoustic parameters of prosody.

4:45

4pSP14. Final stop devoicing in Polish: Incomplete neutralization. Bożena Tieszen and Charles Read (Dept. of Linguistics, Univ. of Wisconsin, 1220 Linden Dr., Madison, WI 53706)

In Polish, as in some other Slavic and Germanic languages, word-final obstruents are devoiced. Three monolingual speakers in Krakow recorded a sample of nine minimal pairs containing final /p,t,k,b,d,g/ preceded by each of the vowels /i,a,u/ in a meaningful carrier sentences. Each word was followed by a voiceless obstruent, in one case, and by a vowel in another. The speakers had also recorded some of the same words in the same environments in a narrative passage. Three durations were measured: Of the preceding vowel, of the stop closure, and of glottal pulsing into the closure. For all three speakers and all environments, the durations of the preceding vowel and of the stop closure did not differ significantly between underlyingly voiced and underlyingly voiceless stops. However, for all three speakers and all three places of articulation, glottal pulsing into the closure was significantly longer for underlyingly voiced stops than for their underlyingly voiceless counterparts. This outcome held for the words in the narrative passage as well as those in the carrier sentences. These results contrast with those obtained by Slowiaczek and Dinnsen [J. Phon. 13, 325-341 (1985)].

Session 4pUW

Underwater Acoustics: Signal Processing II

Nicholas C. Makris, Chair
Naval Research Laboratory, Washington, DC 20375

Chair's Introduction—12:55

Contributed Papers

1:00

4pUW1. Serpentine array processing. J. Robert Fricke (Dept. Ocean Eng., MIT, Rm. 5-218, 77 Massachusetts Ave., Cambridge, MA 02139)

Often one must determine the wave-number spectra of complicated spatial wave fields, which contain significant energy propagating in a broad range of directions. In this case a single linear array must be steered to endfire to compute some of the wave numbers of interest. To alleviate this need a multilinear array, e.g., a Mill's Cross, is often used. This paper discusses a broadband wave-number spectral estimation methodology wherein the array elements may be placed in an arbitrary serpentine pattern. The array processing technique is a modified radon transform for serpentine arrays, which may be interpreted as a plane wave decomposition. The elements of the serpentine array are projected onto virtual linear arrays steered in the intended direction for plane-wave estimation. A weighting function is computed to compensate for multiple serpentine array segments being projected onto a single virtual array segment. The compensation varies with each steering angle. The length of the virtual linear array at each steering angle determines the resolution of the serpentine array. [Work supported by ONR.]

1:15

4pUW2. Large sets of frequency hopped codes with nearly ideal orthogonality properties. Scott T. Rickard and J. Robert Fricke (Dept. Ocean Eng., MIT, Rm. 5-218, 77 Massachusetts Ave., Cambridge, MA 02139)

Significant performance improvements would be realized in many sonar systems if only it were possible to have multiple pings in the water at any given time. The use of orthogonal frequency hopped codes is one way to achieve multiple, simultaneous access. These codes can be designed to have nearly ideal autoambiguity and low cross-ambiguity properties. In some applications it is desirable to use codes drawn from a very large set, say $O(10\,000)$. This paper discusses how such a set can be generated. Specifically, the trade-off between auto/cross ambiguity properties, time-bandwidth product, and code set size when generating large sets are explored. For codes with N frequency hops, the focus is on techniques for generating more than $O(N)$ codes, which characterizes most code generation techniques. Others are nonlinear in N . In particular, the Golomb-Costas method generates $O(N^2)$ codes. These codes are "full", which means every available frequency is used in every code. The Reed-Solomon method, in contrast, generates $O(N^k)$ codes, where k is a trade-off parameter proportional to sidelobe level in the cross-ambiguity function. Reed-Solomon codes, however, are not "full." The trade-off between fullness and number of codes is illustrated. [Work supported by C. S. Draper Laboratory.]

1:30

4pUW3. Effective vertical aperture for three-dimensional arrays in shallow waters. Christopher W. Bogart and T. C. Yang (Naval Res. Lab., Washington, DC 20375)

In shallow waters, the effective vertical aperture of an array composed of horizontal and vertical segments may be greater than the water column depth. The extended vertical aperture leads to better decomposition of bottom-interacting normal modes and hence better range-depth localization compared with the performance of the vertical subarray alone. The conditions under which the effective vertical aperture gives improved localization performance over a vertical array spanning the water column are demonstrated. The effect of source frequency, ratio of horizontal to vertical array length, and bottom properties on localization performance are discussed.

1:45

4pUW4. Digital acoustic video image transmission system for deep-sea research submergence vehicle "SHINKAI 6500." Toshio Tsuchiya, Yasutaka Amitani, Hiroshi Ochi (Deep Sea Technol. Dept., Japan Marine Sci. and Technol. Ctr., 2-15 Natsushima-cho, Yokosuka, 237 Japan), and Michiya Suzuki (NEC Corp., Fuchu, 183 Japan)

High-speed video image transmission from a research submersible to the mother ship is desirable to facilitate effective deep-sea research by a submersible. Therefore, a digital acoustic image transmission system was developed, and was installed in the manned deep-sea research submergence vehicle "SHINKAI 6500" of the Japan Marine Science and Technology Center. This system which adopts 4-DPSK modulation with a carrier frequency of 20 kHz, has a communication capability of 16 kbit/s and is able to transmit a color-still image of 256×240 pixels in 8 s. Its communication range is more than 6800 m. In order to transfer a high-quality color image in a short time, (a) a wide band transducer, that doubles the transmission band, (b) a DCT- (discrete cosine transform) based image coding technique that reduces transmission data volume, and (c) an adaptive equalizer and phase controller, that improve communication quality and Doppler compensation, are employed. Results of a performance test (various video images) were successfully received from the "SHINKAI 6500" from a depth of 6500 m to the surface vessel. After a successful performance test, this system will probably be installed in the "SHINKAI 6500," and new biological or geological experiments will be carried out by the system in the deep sea of the western Pacific Ocean.

2:00

4pUW5. Detection of a submerged object insonified by beach noise. Nicholas C. Makris, William A. Kuperman (Naval Res. Lab., Washington, DC 20375), and Frank Ingenito (SACLANT Undersea Res. Ctr., La Spezia, Italy)

In a previous paper, numerical simulations were performed which demonstrated that a submerged object insonified by surface noise in an ocean waveguide can be detected at low to mid frequency for realistic arrays within about 1-km range [Makris *et al.*, *J. Acoust. Soc. Am.* **92**, 2416 (A) (1992)]. For the present paper, an investigation into the possibility of detecting a submerged object insonified by low- to mid-frequency beach noise in an ocean waveguide was carried out. Model

beach noise is defined as a one-dimensional array of uncorrelated surface sources and the detectability of the object is investigated for various orientations of the sensing array axis, beach axis, and object.

2:15

4pUW6. Time compression processing of M sequences from the Heard Island Feasibility Trial. Garry J. Heard (Defence Res. Establishment Pacific, Bldg. 199, FMO Esquimalt, Victoria, BC V0S 1B0, Canada) and Ian Schumacher (JASCO Res. Ltd., Sidney, BC V8L 3S1, Canada)

Canada was one of nine participating countries in the Heard Island Feasibility Test (HIFT) and had two towed-array receivers operating, one in the Pacific and one in the Atlantic. Due to the extremely long propagation paths the Canadian data were found to have relatively low signal-to-noise ratios that complicated the M -sequence *time-compression* correlation processing. An analysis was carried out to determine if time-compression processing of the data was possible with the low signal-to-noise ratio data available. A means of removing Doppler variations from the data based on phase tracking the heterodyned baseband signal was developed and applied to several of the stronger HIFT receptions. Useful correlation results were obtained using this technique. Initial estimates of the travel time differences for the dominant acoustic arrivals were obtained from the correlations.

2:30

4pUW7. Noise cancellation, environmental focusing, and source localization. Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

The ability to localize an acoustic source in the ocean is often limited by ambient noise and other types of noise. If the nature of the noise is at least partially understood, this limitation can be reduced significantly by including the noise parameters in the space of search parameters. The performance of this approach has been tested in simulations involving a source buried in ambient noise and interference noise due to a source at a known location. Uncertainty in the ocean environment is another limiting factor that can be reduced by expanding the space of search parameters [Collins and Kuperman, *J. Acoust. Soc. Am.* **90**, 1410–1422 (1991)]. By including the source, noise, and environmental parameters in the search space, it is possible to localize an acoustic source buried in noise in an uncertain environment. Special cases of this optimization problem include source localization, tomography, geoaoustic inversion, focalization, and imaging with acoustic daylight [Buckingham *et al.*, *J. Acoust. Soc. Am.* **91**, 2318(A) (1992)].

2:45

4pUW8. Theoretical signal-to-noise gain formulas for energy signal bicorrelation and tricorrelation detection. Lisa A. Pflug (Naval Res. Lab., Stennis Space Center, MS 39529-5004), George E. Ioup (Univ. of New Orleans, New Orleans, LA 70148), Robert L. Field (Naval Res. Lab., Stennis Space Center, MS 39529-5004), and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA)

In the past the authors have examined energy signal detection and time delay estimation performance of ordinary and higher-order correlations using Monte Carlo simulations and hypothesis testing [for example, Pflug *et al.*, *J. Acoust. Soc. Am.* **91**, 2763 (1992)]. For some circumstances higher-order detector performance and SNR gains can be predicted theoretically based on third- and fourth-order moments. SNR gain formulas for higher-order detectors have been derived for a probability of detection of 0.5 and are discussed in this paper. Comparisons of these theoretical results are made to simulations with a model signal whose moments can be varied. Validity of the theoretical results is confirmed with eight diverse test signals which have been used in previous simulation studies. The formulas make possible the calculation of minimum third, and fourth-order moment values needed for the bicorrelation and tricorrelation detectors, respectively, to outperform the cross-correlation detector. [Research supported by Office of Naval Research, Program Element 61153N, with technical management provided by NRLSSC.]

search, Program Element 61153N, with technical management provided by NRLSSC.]

3:00–3:15 Break

3:15

4pUW9. Signal-to-noise improvement for higher-order correlators with one-dimensional prefiltering. Lisa A. Pflug (Naval Res. Lab., Stennis Space Center, MS 39529-5004), Juliette W. Ioup, George E. Ioup (Univ. of New Orleans, New Orleans, LA 70148), and Robert L. Field (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Previous work by the authors has shown that prefiltering when used in conjunction with higher-order correlators possesses potential detection advantages over the ordinary cross-correlation detector for band-limited transient signals in noise for active (known source) and passive (unknown source) sonar. Preliminary simulation results have been presented [Ioup *et al.*, *J. Acoust. Soc. Am.* **92**, 2417(A) (1992); Ioup *et al.*, Proceedings of the IEEE Signal Processing Workshop on Higher-Order Statistics, 7–9 June 1993]. Additional research has been completed, including a systematic and quantitative study using three different filter levels. Eight test signals are presented in the study: Several very low-frequency signals, a pulse, a 50-Hz sinusoid, a resonating sinusoid, a FM linear sweep, and a nonlinear FM sweep. SNR gain comparisons between the detectors are made for $P_d=0.5$ and $P_{fa}=0.001$. Significant SNR gains are evident for both the known and unknown source models with the tricorrelation detector exhibiting the largest gains. For all the test signals, the tricorrelation detector outperforms the cross-correlation detector for the unknown source model, and outperforms the matched filter for seven of the eight test signals for the known source model. [Research supported by Office of Naval Research, Program Element 61153N, with technical management provided by NRLSSC.]

3:30

4pUW10. Minimum entropy deconvolution of underwater acoustic transients. Michael K. Broadhead (Naval Res. Lab., Code 7173, Stennis Space Center, MS 39529-5004)

The minimum entropy deconvolution (MED) method [R. Wiggins, *Geoexploration* **16**, 21–35 (1978)], is used to produce source signature estimates from underwater acoustic transients, which has application in the passive sonar classification problem. The Green's function, representing environmental distortion effects, is treated as a non-Gaussian random process. The MED method implemented in this paper leads to a one-parameter model that doesn't require accurate knowledge of the source location and environmental parameters, or the computation of Green's functions. Results from applying the MED method to single-sensor, broadband ocean acoustic measurements are shown, where source signature estimate correlation coefficients range between 0.87 to 0.91. The method is also applied to realistic simulations, produced with a parabolic equation (PE) model, using a source pulse that was longer and more complex than the measured source. Source signature estimate correlation coefficients ranged between 0.95 and 0.99. The technique assumes that the input random process is non-Gaussian, which is justified by high kurtosis values of the computed (PE) Green's functions associated with this data set. [Work supported by Office of Naval Research, with technical management provided by the Naval Research Laboratory.]

3:45

4pUW11. The effects of a bottom-limited ocean on signal kurtosis—Applications to tricorrelation detection and minimum entropy deconvolution of broadband signals. R. L. Field and J. H. Leclere (Naval Res. Lab., Code 7173, Stennis Space Center, MS 39529-5004)

As signals propagate in the ocean their temporal and spectral features vary as a function of space and time. This is especially true in

bottom-limited environments where the signal may encounter numerous boundary interactions with significant loss due to absorption. This environmental distortion can have a significant effect on the ability to detect and classify broadband signals. A broadband parabolic equation model is used to predict signal distortion over a 25–150-Hz bandwidth in both range-independent and downslope, bottom-limited environments. Signal kurtosis is used to characterize signal distortion in range and depth. Comparisons are made between the kurtosis predicted by the model and that estimated from experimental data. It is shown that the model, together with good environmental information, can accurately predict signal kurtosis as a function of range and depth in the ocean. The significance of the variation of signal kurtosis is discussed in terms of the tricorrelation detector and minimum entropy deconvolution of broadband signals. [Work supported by Office of Naval Research, Program Element 61153N, with technical management provided by the Naval Research Laboratory.]

4:00

4pUW12. Cross-spectral matrix estimation effects on adaptive beamforming. David E. Grant, Jonathan H. Gross, and Mimi Z. Lawrence (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713)

Adaptive beamforming algorithms, which use CSM (cross-spectral matrix) data to calculate optimum weights, are sensitive to the number of unsmoothed CSM samples used in the CSM estimate. Assuming that the samples used in the CSM estimate are independent identically Gaussian distributed with zero mean, it is shown that the output power from the estimated CSM is biased low relative to the output power from the real CSM. Using data from a horizontal line array deployed in a shallow water, high noise environment, the adaptive beamformer output power with various numbers of samples used in the CSM estimate has been measured. The number of CSM samples was varied by changing the time–bandwidth product. The measured median beam noise levels are compared with the theory. Since the data do not fit the assumptions stated above, the agreement is not exact. However, the general trends of the data and theory agree quite well. Conclusions are drawn about the number of samples that should be used for a CSM estimate to be reliable, and about the stationarity of the data within the time-bandwidth integration used in the CSM estimates. [This work is supported by the U.S. Navy Space and Naval Warfare Systems Command.]

4:15

4pUW13. Results on the identification of acoustic objects in motion from the fourth-order cumulant spectrum. Roger F. Dwyer (Naval Undersea Warfare Ctr., New London, CT 06320)

It was reported [J. Acoust. Soc. Am. **93**, 1460–1465 (1993)] that the Doppler component and both the magnitude and phase of an object's

transfer function could be extracted from the fourth-order cumulant spectrum. Results for three elastic spherical objects in motion will be discussed. Specifically, performance results for the extraction of the Doppler component, phase, and magnitude of the object's transfer function from the fourth-order cumulant spectrum will be compared with classical first- and second-order methods.

4:30

4pUW14. Shallow water reverberation, signal motion loss, and acoustic navigation: A preliminary study for autonomous underwater vehicle target detection. W. Kenneth Steward, Xiaou Tang, and Dezhong Chu (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

It is usually difficult to use autonomous underwater vehicles (AUV) for locating targets in shallow-water ocean environments due to the strong wave surge and multipath acoustic reverberations. A preliminary study of some environmental effects unique for a shallow-water sound channel helps place bounds on possible solutions and reduce program uncertainty. Accordingly, three interrelated issues are addressed in this paper: Signal masking by shallow-water reverberation, signal loss caused by extreme platform motions, and shallow-water acoustic navigation. A model estimating the signal motion loss and the average reverberation intensity in the shallow-water environment is developed. The computer simulation of multisensor navigation serves not only as a precise vehicle-positioning device but also as the platform motion data input for the model. As a result of the shallow-water experiments and acoustic modeling, conclusions of this preliminary study are (1) SML is not the dominant factor for sonars in the frequency range of interest (> 200 kHz) with respect to target detectability, (2) beam patterns can be manipulated to effectively reject most interferences caused by surface reverberation and noise, and (3) a multisensor navigation filter is practical even with limited sensor data quality.

4:45

4pUW15. Spatial coherence measurement of sound in the northwest Pacific Ocean. Dinghua Guan, Ruichao Zhu, Renhe Zhang, and Yaoming Chen (Inst. of Acoust., Academia Sinica, P.O. Box 2712, Beijing 100080, People's Republic of China)

An experiment on the transverse horizontal spatial coherence of sound propagating in the ocean was performed jointly by Chinese and Russian acousticians in the northwest Pacific Ocean in June 1990. Three hydrophones with spacings of 270 and 130 m were put in the water at a 30-m depth. The acoustic source with four cw (from 109 to 860 Hz) and a broadband pseudorandom noise signal was drifted at a depth of 100 m. Part of the measurement results of up to 140 km are presented in this paper. It seems that the spatial coherence was related to the amplitude of the received signals and rises considerably in convergence zones.

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with the

U.S. Technical Advisory Group for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics**G. S. K. Wong, Chair S1***Institute for National Measurement Standards (INMS), National Research Council, Ottawa, Ontario K1A 0R6, Canada***H. E. von Gierke, Chair***U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics, 1325 Meadow Lane, Yellow Springs, Ohio 45387***V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics***National Institute of Standards and Technology (NIST), Building 233, Room A149, Gaithersburg, Maryland 20899*

Standards Committee S1 on Acoustics. Working group chairs will report on their preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, will also be discussed. The chairs of the respective U.S. Technical Advisory Groups for ISO/TC 43 (H. E. von Gierke), and IEC/TC 29 (V. Nedzelnitsky), will report on current activities of these Technical Committees. Reports will be given on the results of the last methods of ISO/TC 43 (in Oslo, Norway, held from 31 May–4 June 1993), and of IEC/TC 29 (in Oslo, Norway, held from 24–28 May 1993).

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance, and comfort.

THURSDAY AFTERNOON, 7 OCTOBER 1993

SAVOY ROOM, 3:00 P.M.

Meeting of Accredited Standards Committee S3 on Bioacoustics

to be held jointly with the

U.S. Technical Advisory Group (TAG) Meetings for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock**J. D. Royster, Chair S3***4706 Connell Drive, Raleigh, North Carolina 27612*

H. E. von Gierke, Chair U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics and ISO/TC 108/SC4, Human Exposure Mechanical Vibration and Shock
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics*National Institute of Standards and Technology (NIST), Building 233, Room A149, Gaithersburg, Maryland 20899*

Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock, will also be discussed. The Chairs of the U.S. Technical Advisory Groups for ISO/TC 43

(H. E. von Gierke), and IEC/TC 29 (V. Nedzelnitsky), will report on current activities of these Technical Committees and Subcommittees. Reports will be given on the results of the last meetings of ISO/TC 43 (in Oslo, Norway, from 31 May–4 June 1993), and of IEC/TC 29 (in Oslo, Norway, from 24–28 May 1993). A report will be given on the ISO/TC 108/SC4 meeting held in London, United Kingdom, from 29 March–1 April 1993.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical shock and physiological acoustics, including aspects of general acoustics, shock, and vibration which pertain to biological safety, tolerance and comfort.

THURSDAY EVENING, 7 OCTOBER 1993

GRAND BALLROOM D/E, 7:30 TO 8:30 P.M.

Session 4eAB

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Panel Discussion on Effects of Noise on Marine Mammals

William C. Cummings, Chair

Tracor, Inc., 9150 Chesapeake Drive, San Diego, California 92123-1003

FRIDAY MORNING, 8 OCTOBER 1993

MAJESTIC BALLROOM, 10:00 A.M. TO 12:00 NOON

Session 5aMU

Musical Acoustics: Theatrical Performance

Ronald C. Scherer, Chair

Recording and Research Center, Denver Center for the Performing Arts, 1245 Champa Street, Denver, Colorado 80204

Invited Paper

5aMU1. Contrasting speech and singing—Performance demonstration. Ronald C. Scherer, Tony Church (Denver Ctr. for the Performing Arts, 1245 Champa St., Denver, CO 80204), and Renee Skravanos Root (Univ. of Colorado, Denver, CO 80204)

Nonamplified voice and speech for the stage imposes different production requirements than during conversational speech. Tony Church, noted Shakespearean actor and Dean of the National Theatre Conservatory, and Renee Skravanos Root, operatic and recital artist and singing teacher, will demonstrate voice and speech production for the stage relative to contrasts of communication intent and requirements for audibility and intelligibility. Open discussion and research connections will accompany the demonstrations.

Session 5aNS

Noise: Low-Frequency Sound, Propagation, and Other Noise Topics

Daniel L. Johnson, Cochair

EG&G Special Projects, P.O. Box 9100, Albuquerque, New Mexico 87119

Robert D. Bruce, Cochair

CSTI, 15835 Park Ten Place, Suite 105, Houston, Texas 77084-5131

Chair's Introduction—8:40

Contributed Papers

8:45

5aNS1. Crosswind effects on acoustic propagation. Mark Sprague, Richard Raspet (Phys. Acoust. Res. Group, Univ. of Mississippi, University, MS 38677), and V. E. Ostashev (Inst. of Atmospheric Phys., Russian Academy of Sci., Pyzhevskii 3, Moscow 109017, Russia)

Ray theory says that a ray moving in a crosswind will drift out of the plane of propagation, and that this effect can be significant with high windspeeds and over long propagation distances. White and Li [Michael J. White and Y. L. Li, *J. Acoust. Soc. Am.* **92**, 2405 (A) (1992)] and Wilson [D. Keith Wilson, *J. Acoust. Soc. Am.* **92**, 2405 (A) (1992)] developed two-dimensional fast-field programs (2DFFPs) to calculate the sound pressure levels of windy atmospheres. They reported that crosswind effects were negligible for the cases they considered and that a one-dimensional stationary phase solution yields accurate results. Ray analysis of the two-dimensional phase space used in the 2DFFP indicates that there is a small region that contributes significantly to the sound level. In the presence of a crosswind, the one-dimensional stationary phase approximation does not include all of this region. 2DFFP calculations for realistic windspeeds that show crosswind drift are presented, and the 2DFFP calculations are compared to the calculations of a one-dimensional stationary phase approximation.

9:00

5aNS2. Low-frequency sound waves associated with avalanches, atmospheric turbulence, severe weather, and earthquakes. Alfred J. Bedard, Jr. (Nat'l. Oceanic and Atmospheric Admin., Wave Propagation Lab., 325 Broadway, Boulder, CO 80303)

Recent measurements of naturally occurring atmospheric sound waves near 1 Hz indicate potential uses for monitoring and studying a variety of geophysical processes including avalanches, atmospheric turbulence, severe weather, and earthquakes. A review presents typical signal characteristics including spectral content, showing clear differences between signals from various sources. Possible generation mechanisms are discussed in the context of these data and possible potential practical uses indicated.

9:15

5aNS3. Flare noise damage potential. C. Moritz and R. Bruce (Collaboration in Sci. and Technol., Inc., 15835 Park Ten Pl., Ste. 105, Houston, TX 77084-5131)

Large elevated refinery flares produce significant acoustic power during periods of heavy flaring. Much of this sound power is low frequency, typically in the 4- through 16-Hz octave bands. The intense low-frequency sound can excite resonance frequencies in houses leading to noticeable vibrations of the walls. Through experimental investigations and published data, correlations between hydrocarbon and steam

injection flow rates and far-field sound pressure levels were determined. In addition, correlations between sound pressure levels and typical residential wall vibration levels were also similarly determined. From these relationships, the potential for flare noise-induced vibrations to cause cosmetic damage to nearby houses will be discussed.

9:30

5aNS4. Low-frequency noise-induced vibration of housing structures. Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Tpk., Vernon, CT 06066)

It is known that moderate levels of very low-frequency noise (less than 20 Hz) can induce vibration in elements of common residual housing structures. Measurements of low-frequency noise caused by nearby large industrial facilities were made at several different types of houses. These data are compared to criteria for the perceptibility of noise-induced vibration. The subjective perception by residents of house vibration, and the independent observations of others, correlate well with the noise criteria. In addition, measurements of house vibration were made at some test locations and are compared to both noise and vibration criteria.

9:45

5aNS5. The Lookup program for prediction of noise levels outdoors. Michael J. White and Y. L. Li (U.S. Army Construction Eng. Res. Labs., P.O. Box 9005, Champaign, IL 61826-9005)

A rigorous calculation of the noise level away from a broadband source of known characteristics under known meteorological conditions requires considerable computational effort. However, many practical situations fall into easily described classes of meteorological condition, source and receiver height, ground type, etc. Results have been of tabulated full-wave acoustic calculations, systematically changing the source and receiver configuration, frequency, and environmental parameters. The Lookup program accepts a one-third octave source spectrum, source and receiver configuration, ground condition, and meteorological profile, then accesses the Lookup table for the nearest case and estimates the receiver spectrum. It does this in under 1 s on a PC.

10:00

5aNS6. SoundProp—Fast, accurate prediction of sound propagation under varying weather conditions and over hard or soft surfaces. Paul D. Schomer and Michael J. White (U.S. Army Construction Eng. Res. Labs., P.O. Box 9005, Champaign, IL 61826-9005)

LOOKUP provides a fast, accurate callable sub-routine to predict the one-third octave band received sound spectrum for an arbitrary combination of source height, receiver height and range, one-third-

octave-band source spectrum, and surface impedance (soft or hard). SoundProp is a shell that facilitates the use of LOOKUP. With SoundProp, the user can accurately and quickly perform virtually any sound propagation prediction for a soft or hard surface and a range of sound propagation conditions. Simple barriers can also be included. Future enhancements will include separate specification of the source, path, and receiver surface impedance and additional sound-speed profiles in the LOOKUP library. SoundProp is limited to simple ASCII text file output and must have a VGA monitor. Currently, SoundProp uses about 6 MB of disk space and must be located in drive C. WINSoundProp will be a Windows version of SoundProp and will be able to plot graphical output on any printer supported by Windows. These programs, developed by the U.S. Army Construction Engineering Research Laboratories, are free.

10:15

5aNS7. Finite element and finite difference models for the fast-field program in outdoor sound propagation. Kenneth E. Gilbert and Scott D. Hansen (Appl. Res. Lab. and the Graduate Program in Acoust., Penn State Univ., P. O. Box 30, State College, PA 16804)

The so-called "fast-field program" (FFP) has become a popular method for predicting sound pressure levels in outdoor sound propagation. The method requires the solution of a one-dimensional Green's function equation. The usual approach in outdoor sound propagation has been to obtain the solution in terms of sinusoids or special functions. Here, a simple direct method is demonstrated that uses either finite elements or finite differences in much the same way as does the parabolic equation method. The finite element method presented here is adapted from a range-dependent FFP model developed for underwater acoustics [K. E. Gilbert and R. B. Evans, in *Ocean Seismo-Acoustics*, edited by T. Akal and J. M. Berkson (Plenum, New York, 1986)]. The finite element FFP has been used successfully for propagation problems in atmospheric acoustics for several years. The finite difference approach, which has been developed only recently, is more straightforward than finite elements but equally accurate. We compare the FFP solutions in terms of speed and accuracy using several existing benchmark problems as test cases.

10:30

5aNS8. Computation of turbulent boundary layer wall pressure spectra. Dean Capone (Naval Surface Warfare Ctr., Acoust. Res. Detachment, Code 752, Bayview, ID 83803) and Gerald C. Lauchle (Penn State Univ., State College, PA 16804)

The suitability of various wave-vector-frequency spectral models for predicting the point wall pressure spectrum due to developed turbulent boundary-layer flow over planar, rigid surfaces is investigated. The incompressible and compressible models developed previously [D. Chase, *J. Sound Vib.* **112**, 125-147 (1987)] are given particular attention. The effect of finite-sized measurement transducers is included in the integration of the theoretical wave-vector-frequency spectrum over the in-plane wave numbers in order to arrive at an attenuated point pressure frequency spectrum that can be compared directly to existing experimental data [G. Schewe, *J. Fluid Mech.* **134**, 311-328 (1983) (air), G. C. Lauchle and M. A. Daniels, *Phys. Fluids* **30**, 3019-3024 (1987) (glycerine), and M. P. Horne and R. A. Handler, *Exp. Fluids* **12**, 136-139 (1991) (water)]. The selected experiments cover a wide range of fluid properties and Reynolds numbers: it is found that the optimum set of empirical constants needed to exercise the Chase models depends on the experiment considered. The need to consider viscous effects in future modeling is borne out when dealing with viscous fluids like glycerine. [Work supported by various U.S. Navy projects at NSWC and ARL Penn State.]

5aNS9. Acoustic intensity measurements in the presence of low Mach number turbulent flow. Toby McNeal (Naval Surface Warfare Ctr., Acoust. Res. Detachment, Code 752, Bayview, ID 83803) and Gerald C. Lauchle (Penn State Univ., State College, PA 16804)

Acoustic intensity measurements using the two-sensor cross-spectral density technique are susceptible to errors when made in the presence of mean flow. These errors are predominantly due to turbulent pressure fluctuations acting on and near the sensors. To quantify these errors, intensity measurements were performed in a sound field that contained both an independent acoustic source and a mean, low Mach number ($M < 0.1$), turbulent boundary-layer flow. The microphones were flushed mounted in the wall supporting the turbulent boundary layer and the acoustic source was positioned a large distance downstream. The streamwise component of intensity was measured with and without the flow keeping the sound output of the source constant. Associated bias errors were calculated in accordance with a theoretically established bias error [G. C. Lauchle, *Noise Control Eng. J.* **23**, 52-59 (1984)]. As predicted theoretically, the calculated bias error decreases rapidly as the sensor separation distance increases relative to the correlation length associated with the turbulent boundary-layer pressure fluctuations. [Work supported by various U.S. Navy projects at NSWC and ARL Penn State.]

11:00

5aNS10. Active control of liquid-borne energy in a pipe with a uni-directional wave generator. Lisa Louie (Carderock Div., Naval Surface Warfare Ctr., Carderock, Bethesda, MD 20084-5000)

Active attenuation of narrow-band sound in a liquid-filled steel pipe was conducted experimentally. The pipe is connected to a noise source at one end and opened into a large reservoir at the other end. For active cancellation purposes, a pair of flush-mounted actuators are located in the mid-section of the pipe. The pair of actuators are used to send a uni-directional wave that is equal in amplitude but 180 deg out of phase to the propagating wave down the pipe. The consequence of implementing this technique is that there is no increase of energy between the noise source and the actuators. Results are recorded by hydrophones located in many locations along the pipe and in the reservoir. Over 20-dB reduction was achieved. [Work supported by ONR.]

11:15

5aNS11. Active noise control applied to extended sources in a half-space. Steve Shepard and Kenneth A. Cunefare (The George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Recent work adapted free-field boundary element method-based optimal active noise control techniques for application in a half-space with a rigid or pressure release boundary. Here the required control source strengths and relative phases for active noise control applied to a rectangular solid above a rigid plane are investigated. It is assumed that one face of the solid is radiating harmonically with known surface normal velocity. The existence of unique behavior, as compared to free-field conditions, for the control source strengths and achievable power reduction due to the presence of the plane is demonstrated. This unique behavior occurs at and near certain discrete frequencies. These frequencies are not the characteristic frequencies of the Helmholtz integral for the rectangular solid; rather, they are related to the particular noise source velocity distribution and control source distribution. Finally, the half-space control formulation is employed to evaluate data from an experiment conducted in a hemi-anechoic chamber.

Session 5aPA

Physical Acoustics: Propagation and Nonlinear Acoustics

Charles E. Bradley, Chair

Berkeley Center for Sensors and Actuators, University of California, Berkeley, California 94720

Contributed Papers

8:00

5aPA1. Dispersive pulse propagation and group velocity. Charles E. Bradley (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

The propagation of pulses in a dissipative medium is investigated both theoretically and experimentally. The theoretical work is based on the dissipative dispersion integral, and the measurements are made in an air-filled periodic waveguide (i.e., the dispersion is Bloch wave dispersion). The dispersion integral is considered in the context of a sequence of characteristic pulse duration distances. The pulse propagates without distortion up to the smallest characteristic distance, and thereafter undergoes a new variety of distortion as it encounters each subsequent characteristic distance. Several new solutions of the dispersion integral that exhibit a variety of novel propagation properties are found. Pulses that shift in frequency as they propagate, accelerate as they propagate, and propagate at near-infinite group velocity are found analytically and verified experimentally. [Work supported by ONR.]

8:15

5aPA2. Pulse-splitting in a nonlinear waveguide. Andrés Larraza, William F. Coleman, and Anthony A. Atchley (Phys. Dept., Code PH/La, Naval Postgraduate School, Monterey, CA 93943)

A nonlinear effect in an acoustic duct is described for waveguide modes above the first cutoff frequency, whereby a finite extent modulation of the nonlinear mode splits into two disturbances moving with two different velocities of propagation. That is, for each bit of information input, there will be two bits output. The effect was first predicted by Whitham for nonlinear dispersive media [G. B. Whitham, *Linear and Nonlinear Waves* (Wiley-Interscience, New York, 1974)]. An experiment to verify this phenomenon will be discussed. The apparatus consists of a 21-m-long, 7-segment, thick-walled aluminum tube with a 2-in. internal diameter. Two high-intensity compression drivers arranged in a push-pull configuration are mounted at one end to drive a mode above the first cutoff frequency. A 3-m anechoic termination spanned the length of the last segment at the other end. A physical explanation of the effect, a simplified perturbative treatment of the problem, and possible applications to fiber optic communications will be presented. [Work supported by ONR and NPS Direct-Funding Program.]

8:30

5aPA3. A multidimensional numerical algorithm to simulate the propagation of a shock wave through caustics. Andrew A. Piacsek (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

One strategy in the effort to explain features of observed sonic boom profiles that deviate from the classical N shape is to simulate, one a computer, the propagation of shock waves in an inhomogeneous medium. A time-marching finite-difference code is being developed by the author which employs a multidimensional explicit flux corrected transport scheme, based on work by Zalesak [J. Comp. Phys. **31**, 335-362 (1979)] and McDonald (NPE source code and private communication). Certain essential features of the program have been examined by

comparing the results of some simplified cases to published results of other numerical schemes and, where they are known, to analytical solutions. Specifically, results will be shown regarding the code's ability to simulate the 2-D linear propagation of a curved shock wave front in a homogeneous medium, and 1-D nonlinear propagation of pulses with dissipation and molecular relaxation. Also presented will be some results of 2-D nonlinear propagation, in which an N wave with a curved wave front passes through a caustic. [Work supported by NASA and the William E. Leonhard endowment to the Pennsylvania State University.]

8:45

5aPA4. Propagation of transient waves in a random medium. Alan R. Wenzel (NASA Langley Res. Ctr., MS 460, Hampton, VA 23681-0001)

A theoretical analysis of the propagation of transient scalar waves in a one-dimensional random medium is presented. The index of refraction of the medium is assumed to deviate only slightly from unity, which allows the analysis to be carried out with the aid of a perturbation method. The specific approach adopted here combines a renormalization technique with a travel-time-corrected averaging procedure called asynchronous ensemble averaging. A general expression, valid for an arbitrary initial disturbance, is obtained for the variance of the wave. From that result, an expression for the variance is derived for the special case in which the initial disturbance has the form of a ramp function with arbitrary slope. These results show that the variance of the wave is directly proportional to the variance of the refractive index of the medium, but is only weakly dependent on the propagation path length. It is also found that, as the slope of the ramp function decreases, the wave variance decreases as well. The presentation concludes with some observations on the relevance of these results to the problem of sonic-boom propagation in the atmosphere. [Research supported by NASA.]

9:00

5aPA5. Connection between structural and statistical models of atmospheric turbulence. George H. Goedecke, Paul M. Pellegrino (Dept. of Phys., NMSU, Las Cruces, NM 88003-0001), and Harry J. Auvermann (Army Res. Lab., Battlefield Environment Directorate, WSMR, NM 88002-5501)

A structural model of isotropic homogeneous atmospheric turbulence is presented, in which the turbulence is described as an ensemble of individual localized quasistatic eddies or turbules. Turbules of different sizes a are assumed self-similar with respect to their temperature and flow velocity fields, and randomly positioned and oriented with no correlations. It is shown that, in order to obtain agreement with the Kolmogorov acoustical scattering spectrum, the usual relation between turbule flow speed v and size, $v \propto a^{1/3}$, must be assumed, and, in addition, (i) turbule number densities must be proportional to a^{-3} , and (ii) turbule sizes must be in a geometric sequence, from the outer scale to the inner. It is also shown that the Kolmogorov spectrum is insensitive to the turbule morphology, and that the extent of the "inertial range" is influenced only weakly by the morphology. Expressions for C_7^2 and C_8^2 ,

and for the boundaries of the inertial range, are derived in terms of the structural parameters.

9:15

5aPA6. Comparisons of experimental measurements of sound propagation over a hill and the polar parabolic equation method. Chulso You, Henry E. Bass (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677), and Kenneth E. Gilbert (Penn State Univ., State College, PA 16804)

The polar parabolic equation (POPE) method, which introduces the boundary fitting coordinates into the parabolic equation (PE) method, was developed to solve for sound propagation over a curved surface and over irregular terrain (hills) [X. Di *et al.*, *J. Acoust. Soc. Am.* **92**, 2431 (A) (1992)]. POPE can include realistic sound-speed profiles and ground impedance. Predictions based upon the POPE have been compared to measurements at the JAPE 91 (Joint Acoustic Propagation Experiment) terrain masking experiment at White Sands Missile Range, New Mexico during the period 27–28 July 1991. The sound sources were hovering helicopters at the base of a hill and far away from the hill. In the JAPE data, the relative sound pressure level decreases rapidly from the top of the hill and becomes nearly constant along the base of the hill. This characteristic feature is well predicted by the polar parabolic equation calculations. POPE method appears to be an accurate means for predicting sound propagation over realistic terrain.

9:30

5aPA7. Determination of turbulent velocity correlations by the nonlinear scattering of crossed ultrasonic beams. Murray S. Korman and James E. Parker, III (Dept. of Phys., USNA, Annapolis, MD 21402)

The nonlinear interaction of two, mutually perpendicular crossed ultrasonic beams, overlapping in the presence of turbulence, generates a scattered sum frequency component that radiates outside the interaction region. In the absence of turbulence, virtually no scattered sum frequency component exists (outside the interaction region). A theoretical investigation is reported that relates the shape of the ensemble averaged scattered sum frequency intensity spectrum, $I_+(\omega, \theta_*)$ (which exhibits a Doppler shift, frequency broadening, skewness, and kurtosis), to the scattering angle θ_* , incident and scattered wave vectors (where $\mathbf{K}_+ = \mathbf{k}_s - [\mathbf{k}_1 + \mathbf{k}_2]$), and statistical properties of the turbulent velocity field \mathbf{v} . The n spectral moments $\langle (\mathbf{K}_+ \cdot \mathbf{v})^n \rangle \propto \int I_+(\omega, \theta_*) (\omega - \langle \omega \rangle)^n d\omega$ (obtained from experiment) are used to evaluate turbulent velocity correlations like $\langle v_x v_y \rangle$, where $n=2$. The scattering geometry involves rotating the axis of the transmitting crossed beams (which are always perpendicular to each other) in the plane containing the submerged circular water jet and receiver axes. Angle θ_* is measured between the ray bisecting the transmitting axes and the stationary receiver axis. The crossed beams are focused and overlap at the common focal point. Spectral moments, obtained from scanning the overlap region across the jet, are used to predict velocity correlations across the width of the jet with good spatial resolution.

9:45

5aPA8. Calculation of average turbulence effects on sound propagation based on a fast field program formulation. Richard Raspet and Wenliang Wu (Univ. of Mississippi, University, MS 38677)

Daigle has published a series of papers in which he has applied the turbulent scattering theories of Chernov and Karavainikov to sound propagation over hard and finite impedance grounds. In these papers, Daigle has introduced the decorrelation in phase and amplitude due to turbulence along the direct and reflected path into the spherical wave reflection analysis for a nonrefracting atmosphere. The phase and amplitude decorrelation terms have been incorporated into the evaluation of the spectral integral of a fast field program for propagation in a refracting atmosphere. Although the calculation involves two significant

approximations it reproduces Daigle's results for homogeneous atmospheres and compares well with the upward refraction measurement of Parkin and Scholes and with measurements taken under a variety of refractive conditions at Bondville, Illinois by the U.S. Army Construction Engineering Research Laboratory.

10:00

5aPA9. Examination of sonic boom propagation through turbulence with ray theory. Leick Robinson (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

The properties of sonic boom propagation through turbulence continue to be explored by a ray theory approach. Ray theory is used to model the effects of refraction and subsequent folding of the wave front by an instantaneous realization of the turbulence. Comparison is made with other prediction methods. These results are compared with characteristics observed in sonic boom measurements. The relative contributions of the large and small scale modes of the turbulent field are evaluated.

10:15

5aPA10. Comparison of measured and predicted pure tone propagation levels from JAPE-1: An evaluation of the performance of ASOPRAT. Carl K. Frederickson, Henry E. Bass, Richard Raspet, and John Messer (Phys. Acoust. Res. Group, Univ. of Mississippi, University, MS 38677)

Short-range propagation data (< 2 km) from the Joint Acoustic Propagation Experiment Phase One (JAPE-1) has been used to evaluate the performance of the advanced sound propagation in the atmosphere (ASOPRAT) computer code. Speakers mounted 2 and 30 m above the ground were used as pure tone sources for the short-range measurements. Pressure levels measured in the time domain have been Fourier analyzed to produce plots of the pressure due to a particular frequency over time. Meteorological profiles measured at the experiment site were used as input for the prediction routine ASOPRAT. Agreement between the predicted and measured propagation losses is very good. Source strength data were not available as of the writing of this abstract, so at this time comparisons are all relative to a reference microphone at the ground. Source strength data may be available by the time of the meeting.

10:30

5aPA11. Comparison between short-term and long-term averages of turbulence parameters for use in atmospheric acoustics. John M. Noble (U.S. Army Res. Lab., Battlefield Environment Directorate, ATTN: AMSRL-BE-M, White Sands Missile Range, NM 88002)

Sonic anemometers are good instruments for measuring temperature and wind speed fast enough to calculate the temperature and wind structure parameters used to calculate the variance in the acoustic index of refraction. The problem is how long of an average is needed to obtain a representative value of the strength of the turbulence field in the atmosphere. In addition to this problem, there are several problems associated with making point measurements and using them to represent a turbulence field. These problems will be examined by analyzing some of the sonic anemometer data from the Joint Acoustic Propagation Experiment (JAPE) conducted during July 1991 at DIRT site located at White Sands Missile Range, New Mexico. This experiment provides turbulence data from sonic anemometers at five heights for several hours a day over a period of approximately 2 weeks.

10:45

5aPA12. Application of the CFFP to sound propagation in inhomogeneous media above a ground with an impedance discontinuity. Y. L. Li and Michael J. White (US Army Construction Eng. Res. Lab., P. O. Box 9005, Champaign, IL 61826-9005)

Using the chirp fast field program (CFFP), a systematic method has been developed for numerical computation of sound propagation in an inhomogeneous atmosphere above a ground with an abrupt change in impedance. In situations where a noise is generated above one type of ground surface (say pavement) and travels beyond its edge, onto a second type of surface (e.g., grassy ground), each surface usually controls the rate of attenuation near it. When the air near the ground is refractive, it may emphasize the attenuation during downward refraction and eliminate it in upward refraction. Other effects of the inhomogeneous atmosphere are discussed. Several interesting examples will be presented.

11:00

5aPA13. Low-frequency wind noise for a microphone inside a spherical foam windscreen. Scott Morgan (Dept. of Phys., Southeastern Louisiana Univ., 878 SLU, Hammond, LA 70402) and Richard Raspet (Univ. of Mississippi, University, MS 38677)

Windscreens are commonly used to reduce wind noise in outdoor measurements by shielding the microphone from the incoming flow. In this paper a theoretical model for wind noise reduction of a spherical foam windscreen and experimental evidence supporting this model are presented. Results show that wind noise reduction is approximately independent of windscreen diameter for turbulence scale sizes that are much larger than the windscreen and that wind noise reduction scales well with the screen number (defined as the ratio of the windscreen diameter to the scale size of the turbulent eddies) for screen number values less than one. Calculations using average windscreen surface pressure measurements yield wind noise reductions that are of the same order of magnitude as those measured.

11:15

5aPA14. Sound propagation in inhomogeneous media above a ground with a barrier. Y. L. Li and Michael J. White (U.S. Army Construction Eng. Res. Lab., P. O. Box 9005, Champaign, IL 61826-9005)

To study the problem of scattering of acoustic waves from a ground with a barrier, the exact expression of the Green's function due to a line source embedded in a layered medium has been successfully incorporated into the Helmholtz-Kirchhoff integral equation. An extension of the method of moments, which can solve the Helmholtz-Kirchhoff integral equation for studying the problem of sound scattering in the inhomogeneous medium has been developed. Using the method, the effects of scattering in the region beyond a barrier are studied for situations when the background medium is inhomogeneous. The effects of the inhomogeneous atmosphere are discussed. Several interesting examples will be presented.

11:30

5aPA15. Measurements of focused, finite amplitude sound beams reflected from curved targets. Michalakis A. Averkiou, Inder Raj S.

Makin, and Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

The reflection of focused, finite amplitude sound beams from curved targets in water was investigated experimentally and theoretically. Measurements of small signal diffraction effects were used to characterize the effective radius (1.9 cm) and focal length (15 cm) of the source, the center frequency of which is 2.25 MHz. Detailed comparisons are made between measurements of harmonic generation in the incident focused beam and numerical results from a computer code that solves the KZK equation in the frequency domain [Naze Tjøtta *et al.*, *J. Acoust. Soc. Am.* **89**, 1017-1027 (1991)]. Reflection from both convex and concave target surfaces is considered. The targets are made of nickel with different radii of curvature down to a minimum of 5 cm. Measurements of the reflected beam were obtained with a membrane hydrophone that was placed between the source and the target. Theoretical predictions for harmonic generation in the reflected field were obtained by modifying the computer code to account for the phase shifts introduced by interaction of the incident beam with the target. Theory and experiment are in good agreement for both the incident and reflected beams. [Work supported by the Packard Foundation and the Office of Naval Research.]

11:45

5aPA16. Measurements of finite amplitude pulses radiated by plane circular pistons in water. Michalakis A. Averkiou and Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

Measurements are reported for finite amplitude acoustic pulses radiated by plane circular pistons in water. Pulses with center frequencies of several megahertz, peak sound pressures up to 1 MPa, durations ranging from approximately 2 to 20 cycles, and different amplitude and frequency modulations were investigated. Measurements of short pulses were made very near the source, where the center wave and edge wave can be separated. Pulse envelope distortion that accompanies shock formation in frequency-modulated tone bursts is demonstrated. Acoustic saturation of pulsed sound beams is also investigated. All measurements are compared with theoretical predictions obtained from a computer code that solves the KZK equation in the time domain [Lee and Hamilton, *Ultrasonics International 91 Conference Proceedings* (Butterworth-Heinemann, Oxford, 1991), pp. 177-180]. Very good agreement between theory and experiment is obtained both in the near field and the far field, on and off axis. Artifacts in measurements of waveforms containing shocks, which are attributed to bandwidth limitations of membrane hydrophones, are discussed. [Work supported by the Packard Foundation and the Office of Naval Research.]

Session 5aSA

Structural Acoustics and Vibration: Radiation and Scattering Measurements

Geoffrey L. Main, Chair

Office of Naval Research, 800 North Quincy Street, Arlington, Virginia 22217

Contributed Papers

8:30

5aSA1. Energy branching of a subsonic flexural wave on a plate at an air-water interface: Transition radiation and the acoustic wave field in water. Thomas J. Matula and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

The diffraction of subsonic flexural plate waves due to a discontinuity in fluid-loading is experimentally investigated. A tone burst of flexural waves propagates down a plate, the lower section of which is submerged in water. Observations indicate that there occurs a branching of energy as the flexural wave passes through the air-water interface. A portion of the energy continues along the plate as a subsonic flexural wave with an associated evanescent wave. A second acoustic wave (which is termed transition radiation) originates at or near where the plate crosses the interface, and propagates in water to the far field. In the near field of the interface there exists an interference between the two acoustic waves in water that results in a series of pressure nulls. The pressure nulls are associated with a π phase change in the wave field and are indicators of wave front dislocations [P. L. Marston, "Geometrical and Catastrophe Optics Methods in Scattering," *Physical Acoustics* (Academic, New York, 1992), Vol. 21, pp. 1-234]. A computation of the wave field in an unbounded fluid due to a line-moment excitation of a plate is comparable with the null pattern observed but differs in certain details. [Work supported by ONR.]

8:45

5aSA2. Repulsion of phase-velocity dispersion curves and the nature of plate vibrations. B. Hosten, M. Deschamps, A. Gérard (Lab. de Mécan. Phys., Univ. Bordeaux I, F-33405 Talence, France), and H. Überall (University Bordeaux I and Catholic Univ. of America, Washington, DC 20064)

The Lamb waves propagating in an elastic plate *in vacuo* generate compressional (L) and shear type (T) plate vibrations that are coupled due to the boundary conditions. Without such coupling, their phase-velocity dispersion curves would form two intersecting families, which at high frequency tend towards the elastic-wave speeds C_L and C_T , respectively. It is shown that the coupling causes a repulsion of the dispersion curves, similar to that encountered in atomic physics for the energy levels of atoms combining into molecules, which prevents their intersection and at the same time exchanges the nature ($L \leftrightarrow T$) of the underlying vibrations. However, in the repulsion regions a *succession* of dispersion curves combines to asymptotically approach the uncoupled L or T dispersion curves, respectively. For the case of a plate bounded by fluid on one side, and vacuum on the other, the dispersion curves of the fluid-borne (Stoneley-Scholte type) wave, which is known from the studies of Grabovska and Talmant to be present in this case, and of the usual A_0 Lamb wave exhibit a similar repulsion phenomenon.

9:00

5aSA3. Fresnel array method for determining the dispersion curves in plates. R. Daniel Costley, Jr. (Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

and Yves H. Berthelot (Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Experiments have been conducted on the thermoelastic generation of Lamb waves in plates by a Q -switched ruby laser. Detection was performed both by interferometric laser probe and a miniature piezoelectric pinducer. The long-term objective of this work is to determine various characteristics of the plate such as material properties, plate thickness, flaws, and inhomogeneities, or even bonding between plates by comparing experimentally determined points with the theoretical dispersion curves. The technique consists in forming on the surface of the specimen an array of confocal arc sources by passing the laser beam through a Fresnel lens. The array spacing produces a "forcing wavelength" for which only a few specific frequencies can propagate. Dispersion curves can be obtained by measuring the frequency content of the received signals for a range of wave numbers. Because of the narrow-band nature of the technique, and because of the confocal geometry of the source distribution, this technique offers a relatively high signal-to-noise ratio. Good agreement is obtained between theoretical and experimental dispersion curves especially for the lower modes, thus showing that the proposed technique has potential for the above-mentioned applications. [Work supported by the National Science Foundation.]

9:15

5aSA4. Impulse response for backscattering by a thin spherical shell: Measurement and wave interpretation. Gregory Kaduchak, Christopher S. Kwiatkowski, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A broadband sheet source was developed to produce a pressure impulse with a planar wave front containing a wide range of frequency components [C. S. Kwiatkowski *et al.*, to be presented at this meeting]. This source is nearly acoustically transparent and was used for backscattering from an empty stainless steel spherical shell where prominent features in the shell's calculated impulse response are observed over a wide frequency interval. A wideband hydrophone was placed in the far field of the scatterer on the opposite side of the source. Time records reveal a gaussian wave packet associated with the excitation of the subsonic a_{0-} wave responsible for a large backscattering enhancement near the coincidence frequency. Superposed on the same records are large contributions from the low frequency excitation of the breathing mode [G. Kaduchak and P. L. Marston, *J. Acoust. Soc. Am.* **93**, 2700-2706 (1993)]. The shell used in the experiment has a thickness to radius ratio of 2% for which the above scattering phenomena occur for frequencies less than 450 kHz. [Work supported by ONR.]

9:30

5aSA5. Observation of the prompt high-frequency enhancement of tone bursts backscattered by a thin spherical shell near the first longitudinal resonance. Gregory Kaduchak and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A prominent feature predicted for the backscattering of tone bursts by thin spherical shells is an enhancement of a guided wave contribution

near the first longitudinal resonance. This has been explained with a backward ray model of a leaky Lamb wave where energy is leaked off without having circumnavigated the back side of the shell [P. L. Marston *et al.*, *J. Acoust. Soc. Am.* **90**, 2341 (1991); D. H. Hughes, Ph.D. thesis, Washington State University (1992)]. The relevant s_{2b} Lamb wave has opposing group and phase velocities giving rise to prompt radiation following the direct specular echo. The present research gives a comparison between a ray theory approximation and experiments in which tone bursts having carrier frequencies in the range $595 < ka < 640$ were incident on an empty stainless steel spherical shell in water of radius $a=12.7$ cm. The sphere's thickness to radius ratio is 2%. The response generally follows the predicted amplitude which can be close to 4.5 times the specular amplitude. [Work supported by ONR.]

9:45

5aSA6. The influence of internal structures on bistatic scatter from finite cylindrical shells near axial incidence. Matthew Conti and Ira Dyer (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Previous results have examined the scatter from an empty shell at axial incidence illustrating the primary role of the endcap. Building on this, important differences in the bistatic scatter due to the presence of internal structures are presented. In addition, differences due to source incidence away from axial are investigated. Discussion is based upon bistatic measurements on an empty shell with endcaps, a duplicate shell with internal ring stiffeners and a duplicate ring stiffened shell with resiliently mounted wave-bearing internal structures. Integrated target strengths shown as a function of observation angle reveal spatial regions where the internal structures cause significant change. Of particular interest is enhancement of target strength near backscatter. In conjunction with conventional beamforming techniques, these analyses reveal the distribution of energy along the shell length. This distribution is modified by rotation of incidence away from axial. Relative decay rates and frequency-dependent effects are also discussed. The bistatic measurements were conducted over a frequency range of $2.5 < ka < 10$, corresponding to 3/4 to 3 times the ring frequency of the shells. [The authors acknowledge the assistance of NRL for acquisition of data. Work supported by ONR.]

10:00

5aSA7. Experimental investigation of nonspecular scattering from a framed shell. Brian H. Houston, Joseph A. Bucaro, and Douglas M. Photiadis (Phys. Acoust. Branch, Naval Res. Lab., 4355 Overlook Ave., Washington, DC 20375)

Recently the existence of flexural Bloch waves contributing significantly to features observed in far-field experimental data was reported [Photiadis *et al.*, *J. Acoust. Soc. Am.* **93**, 2413(A) (1993)]. In this previous work, broadband scattering data from framed and unframed low aspect ratio fluid-loaded shells ($l/a=3$) were studied. Calculations of the dispersion curves for an infinite ribbed shell were used to predict the locations of the peaks in the backscattered cross section. Here, the results of the scattering from a high aspect ratio ($l/a=8$) pseudoperiodically framed shell are reported. The various elastic wave types using the infinite framed shell dispersion relations are identified and the relevant contributions are quantified.

10:15-10:30 Break

10:30

5aSA8. Scattering investigation with cylindrical acoustic holography. Ming-Te Cheng and J. Adin Mann III (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50010)

The objective of this paper is to investigate sound scattering of a rigid scatterer using a two-plane cylindrical acoustic holography method. A piston source, which is vibrating sinusoidally, illuminates a scatterer. The superimposed incident and scattered fields are measured in the near field of the scatterer using cylindrical acoustic holography.

These two fields are decomposed in the wave-vector domain in order to extract the scattered field. Numerical simulations are performed and the effect of various parameters is investigated. Specifically, the distance between the two-holography cylinder, the sampling rate, and the aperture size are investigated in the field separation technique. In addition, experimental studies were conducted inside an anechoic chamber with a baffled loudspeaker as a source and a cylinder as a scatterer. The decomposed scattered field is then used to estimate the far-field target strength. The near-field scattering is also investigated.

10:45

5aSA9. Comparison of dual surface and tri-surface approaches for acoustical holography measurements of radiated and scattered fields. Michael A. Sartori and Joseph A. Clark (Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

Using near-field measurements of the pressure field of an immersed structure, an estimate of the structure's radiated field and scattered field can be made. With the pressure and the pressure gradient measured at discrete locations in the near field, a discretized solution of the exterior Helmholtz integral is solved to obtain the far-field pressure. This approach has been shown to work for both acoustic radiation and acoustic scattering problems. Using two normal surfaces to approximate the pressure gradient, parametric simulations were performed to determine the scattered field predictability of the approach for an insonified rigid sphere. The results indicate that this dual surface approach better predicts the scattered field when the surfaces are very close, which implies that a good approximation of the gradient is needed. A different approximation of the pressure gradient uses three surfaces instead of two. Using a cylindrical shaped measurement surface, the radiated pressure field from a monopole and the scattered pressure field from a rigid sphere are simulated for the different cases of the approximated gradient, namely the dual surface approach and the tri-surface approach. The effects on the predictability of the radiated field and the scattered field are discussed. [Work supported by ONR.]

11:00

5aSA10. Wave vector measurements of empirical design transfer functions. Joseph A. Clark and Michael A. Sartori (Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

Empirical design transfer functions can be used to predict the radiated noise generated from various components in a system. The Darby method is one example of an approach used to find these empirical design transfer functions. For an immersed and radiating structure, near-field measurements of the pressure field can be wave vector filtered to find the empirical design transfer function from the structural input point to the acoustic far field. Assuming a cylindrical body excited by a force and radiating, a discrete line array of pressure sensing devices are used to measure the near-field pressure. The acoustic pressure data measured at each point are sampled, digitized, time averaged, and frequency transformed. The position-frequency domain pressure is then transformed into the wave vector domain, and the far-field radiation is predicted. The empirical design transfer function is found by dividing the predicted far-field radiation by the input force. The mathematical development of this approach is presented. Simulations are presented for various radiation sources, including a monopole and a discrete line source. The pressure predictions of the wave vector are compared with the pressure directly computed for the sources. [Work supported by ONR.]

11:15

5aSA11. Nature of the Scholte wave on a cylindrical shell. G. Maze, F. Léon, J. Riposte (LAUE, Univ. Le Havre, 76610 Le Havre, France), and H. Überall (Univ. of Le Havre and Catholic Univ. of America, Washington, DC 20064)

The real parts of the eigenfrequencies of an evacuated, fluid-immersed infinite cylindrical shell can be used to obtain the dispersion

curves versus frequency of the phase velocities of families of circumferential waves. These waves are analogous to those of the families of Lamb waves on an elastic plate *in vacuo*, but contain an additional *A* wave corresponding to the Scholte–Stoneley wave in the fluid loading [see M. Talmant, Ph.D. thesis, Univ. of Paris VII (1987), or M. Talmant *et al.*, *J. Acoust. Soc. Am.* **86**, 278 (1989)], which is approximated by the first Franz wave [B. Clotteau *et al.*, *J. Acoust.* **3**, 213 (1990)]. Near the coincidence frequency, a repulsion phenomenon is obtained between the (Lamb wave) A_0 and the *A*-wave dispersion curves, analogous to that in perturbed, quasidegenerate atomic levels. An analytic explanation is provided, using perturbation theory, for the form of the dispersion curves and the nature of the corresponding waves in the region of repulsion.

11:30

5aSA12. New method of investigation of large bandwidth response of mechanical structures. M. Lagier, G. Vanderborck (Thomson Sintra ASM, 525, Route des Dolines, BP157, 06903 Sophia Antipolis Cedex, France), F. Mortessagne, O. Legrand, and D. Sornette (CNRS URA 190, Faculté des Sciences, 06108 Nice Cedex 02, France and X-RS, Parc-Club, 91893 Orsay Cedex, France)

The experimental measurements of high-order mechanical resonances of metallic structures in the high-frequency domain is generally difficult and unreliable, due to the methods of excitations and analysis. Here, an original technique associating a nondestructive percussive excitation together with an acoustic determination of the near field created by the vibrations of the structure in air are proposed. The excitation is obtained using a steel ball magnetically activated, which creates a calibrated impulse. The observation is made using a large bandwidth microphone. The experimentations have been carried out on “gongs” (aluminum plates of 1-mm thickness with other dimensions of the order of the meter) possessing noncanonical shape (stadium, sarcophagus...). With this high signal-to-noise ratio obtained in an anechoic room, it is possible to obtain several hundred eigenfrequencies with a resolution of the order or better than 1/10th the difference between neighboring frequencies. This was attained without any particular treatment other than the spectral LMS analysis. The signal dynamics is around 80 dB, which allows one to obtain the pole and zeros of the transfer function, implying that the amount of information thus obtained is considerable. The results were then compared with those predicted within the theoretical framework proposed previously, in terms of quantum chaos methods applied to high-frequency vibrations of structures, in order to characterize the nature of the spectrum and the vibration eigenfunctions.

FRIDAY MORNING, 8 OCTOBER 1993

LOWER PLAZA, 8:00 A.M. TO 12:00 NOON

Session 5aSP

Speech Communication: Topics in Speech Communication Research (Poster Session)

James R. Sawusch, Chair

Department of Psychology, Park Hall, State University of New York at Buffalo, Buffalo, New York 14260

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered posters will be at their posters from 8:00 to 10:00 a.m. and contributors of even-numbered posters will be at their posters from 10:00 a.m. to 12:00 noon.

5aSP1. Acoustic and syntactic correlates of hesitations in spontaneous speech. Nanette Veilleux, Ahwat Schlosser, and Mari Ostendorf (ECS Dept., Boston Univ., 44 Cummington St., Boston, MA 02215)

One of the difficulties in modeling the prosody of spontaneous speech is distinguishing hesitations from fluent intonational phrase breaks. The goal of this study is to identify acoustic and syntactic cues that can be used to better model hesitation phenomena. The study is based on a set of several hundred utterances of spontaneous speech from the ATIS corpus, each with prosodic constituents and hesitations labeled by hand and syntactic constituents and phonetic alignments labeled automatically. The first set of analyses compared acoustic correlates for prosodic phrase boundaries with and without hesitations. For constituents below the level of the intermediate phrase, the average duration of a pause is longer when there is a hesitation (mean 270 ms vs 19 ms), as is the mean normalized segment duration in the word-final syllable rhyme (mean 1.8 vs 0.2), both with significance $p < 10^{-5}$. The second set of analyses looked at the differences in syntactic structure at word boundaries with and without hesitations. Again, for constituents below the level of the intermediate phrase, results show that hesitations are likely to occur after a conjunction, auxiliary verb, or preposition, i.e., at locations where the entropy of next-word candidates is relatively high.

5aSP2. Vowel classification using a microsegmental model for formant transition based on locus equations. L. Deng and D. Braam (Dept. of Elec. and Comput. Eng., Univ. Waterloo, ON N2L 3G1, Canada)

In the work by Sussman *et al.* [*J. Acoust. Soc. Am.* **90**, 1309–1325 (1991)], locus equations, which describe linear relationships between the onset and steady-state formant values in consonant–vowel syllables, have been experimentally tested using a large quantity of acoustic data and been proposed as a source of relational invariance for stop place categorization. In this presentation, a statistical model is presented, which utilizes the conceptualization of the locus equations as a basis for parametric modeling of phonetic contexts—place of articulation, and of their acoustic consequences—formant transitions. The model is based on hidden Markov model representation of formant–transmission microsegments of speech. A generalized expectation-maximization algorithm is developed for automatic estimation of the model parameters. The proposed model is capable of generalizing consonant characteristics from a small training data set where the contextual information is only sparsely represented, and is hence applicable to very large vocabulary speech recognition problems. Results from vowel classification experiments (TIMIT database), demonstrating relative performance between this locus-based model and the conventional hidden Markov model, will be presented.

5aSP3. The phonological reality of locus equations across manner class distinctions: Preliminary observations. Harvey M. Sussman (Depts. of Linguist. & Speech Commun., Univ. of Texas, Austin, TX 78712)

Locus equations, regression fits to coordinates obtained by plotting F_2 of a variety of vowels at their midvowel nuclei in relation to F_2 at the acoustic onset of the CV syllable have been shown to phonetically index place of articulation within oral stop categories across speakers of diverse languages—English, Swedish, Cairene Arabic, Thai, and Urdu.

The purpose of this study was to determine if locus equations best encode featural or segmental entities. Several consonantal manner classes were assessed. The inventory consisted of CVt tokens having initial approximants—/w/, /j/, /r/, /l/; fricatives—/v/, /ð/, /z/, /ʒ/; nasals—/m/, /n/. In addition voicing contrasts were examined using voiceless stops (s/C₁/VC₂) with C₁=/p/, /t/, /k/. Ten medial vowel contexts were used. Locus equations were found to acoustically distinguish place contrasts within each manner class, with nasality and voicing not affecting the place feature. However, similar place features across stop–fricative–approximant classes, did not correspond in terms of slope and y-intercept values. It was concluded that locus equations best characterize segment-based rather than feature-based, entities.

5aSP4. Effects of speaking rate, focal stress, and sentence position on spectral characteristics and vowel duration. Dawn M. Behne (Linguist. Inst., Univ. of Trondheim, N-7055 Dragvoll, Norway) and Lynne C. Nygaard (Indiana Univ., Bloomington, IN 47405)

Speaking rate, focal stress, and sentence position can affect the duration and spectral characteristics of vowels [e.g., B. Lindblom, *J. Acoust. Soc. Am.* **35**, 1773–1781 (1963)]. However prior research has been less than conclusive in characterizing effects of these factors [cf. e.g., T. Gay, *J. Acoust. Soc. Am.* **63**, 223–230 (1978)]. Previous findings [D. M. Behne and L. C. Nygaard, *J. Acoust. Soc. Am.* **90**, 2254(A) (1991)] have confirmed the combined effects of these factors on vowel duration and have shown that vowels can be independently affected by speaking rate, focal stress, and sentence position. The goal of this paper is to characterize the effects of these factors on formant frequencies of vowels and relate them to effects on vowel duration. Using the vowels /i, o, a, æ/, conversations were developed with /kVt/ and /kVd/ as target words. In each conversation a target word occurred in initial and in final sentence position and was either focused or nonfocused by the discourse. Twelve subjects produced each conversation at three speaking rates. The present results demonstrated no effect of either speaking rate or focal stress on the spectral characteristics of the vowels, but relatively high first formant frequency and low second formant frequencies for vowels in final sentence position compared to initial sentence position. These findings verify that speaking rate, focal stress, and sentence position can concurrently influence syllable-internal timing with distinct acoustic effects. The results will be discussed in terms of vowel reduction and undershoot.

5aSP5. The influence of stress, vowel, and juncture cues on segmentation. Paul N. Yerkey and James R. Sawusch (Dept. of Psychol., Park Hall, State Univ. of New York, Buffalo, NY 14260)

Cutler and Norric [JEP:HPP **14**, 113–121 (1988)] proposed a segmentation strategy based on strong syllables. According to their strategy, a listener attempts lexical access every time a strong syllable is heard. In a word monitoring task, Cutler and Norris found support for this strategy with speakers of British English. However, in the Cutler and Norris experiments, the strong syllables used were characterized by strong stress, a tense vowel, and the possible presence of allophonic cues to word junction. Thus the relative roles of these cues in segmentation is still to be determined. As reported at previous meetings of the Acoustical Society, Yerkey and Sawusch [*J. Acoust. Soc. Am.* **91**, 2338(A) (1992), *J. Acoust. Soc. Am.* **92**, 2443(A) (1993)] examined this issue by separating syllable stress and vowel color in an attempt to determine their relative roles as cues to segmentation. The results of these experiments suggest that, for American English, syllable stress is a primary cue for segmentation, and vowel color does not seem to have an important influence. The present study extends these results by evaluating the role of allophonic juncture in word segmentation.

5aSP6. Development of the song selectivity by nonlinear neural response interactions in the zebra finch. Dukhwan Lim (Div. of Biol., California Inst. of Technol., Pasadena, CA 91125)

The zebra finch's song control system contains song-selective neurons. One of the song nuclei, hyperstriatum ventral, caudal (HVc) receives auditory inputs from the forebrain auditory area, field L, which is subdivided into L1, L2, and L3. Where and how this selectivity is formed has not been studied quantitatively in this forebrain area. The

song of the zebra finch consists of simultaneous and sequential arrangements of acoustic elements. Neuronal responses to individual song elements and to syllables composed of harmonic frequencies were studied in each subdivision of field L and HVc. The responses to pairwise sequences or harmonic syllables differed from their responses to an individual element or a tone presented in isolation. These nonlinear responses were evaluated by a matrix-based analysis method. Results showed a hierarchical increase of nonlinear interactions as a recording site moved from field L to HVc. Both sequential and simultaneous nonlinear interactions were enhanced from field L to HVc. These interactions involved either inhibitory or facilitatory effects. Although, in theory, this selectivity could be formed by a linear way, the song selectivity proved to be achieved dominantly by these nonlinear interactions. [Work supported by IHFSPO.]

5aSP7. Relative perceptual prominence of fundamental frequency peaks in the presence of declination. Bruno H. Repp (Haskins Labs., 270 Crown St., New Haven, CT 06511-6695), H. H. Rump, and Jacques M. B. Terken (Inst. for Perception Res., 5600 MB Eindhoven, The Netherlands)

Two experiments were conducted to determine the perceived relative prominence of two accented syllables within the same utterance. The fundamental frequencies (F_0) at the two accent peaks (P_1, P_2) were varied as well as the rate of F_0 declination over the unaccented syllables. Extending earlier work by Terken [*J. Acoust. Soc. Am.* **89**, 1768–1776 (1991)] with low-pitched reiterant speech, a real Dutch sentence was employed in both female and male pitch ranges. The results confirm earlier observations that P_2 is usually lower than P_1 when judged to be equal in prominence, and that this difference increases with P_1 height. However, this second effect diminishes as declination rate increases. The results do not fit any simple model of prominence perception based on distance from a reference declination line. [Work carried out at IPO, Eindhoven.]

5aSP8. Aural differentiation of related speakers. Winfried Kriebber and Vigo Rinas [Inst. for Commun. Eng., Aachen Univ. of Technol. (RWTH), Melatener Str. 23, D-52056 Aachen, Germany]

Motivated by the well-known identification problems in telephone communication, this study investigates the influence of filtering on the aural differentiation of related speakers (e.g., father and son). Speech samples of couples of related speakers were presented to 30 listeners in 14 sessions (1 couple/session). Each session started with a reference sample to acquaint the listeners with one speaker of the couple. Subsequently the listeners received samples of both speakers and had to detect the samples belonging to the acquainted speaker. Additionally they gave information about the certainty of their decision using the terms "surely," "probably," or "possibly." The following speech samples were used: voiced vowels and (unknown) sentences, unfiltered and filtered with several bandpass filters; voiced vowels without higher formants (only F_1 left), and whispered vowels. In an additional opinion test the similarity of both voices of each couple was rated. The rates of false identifications depend clearly on the filter bandwidth. For voiced vowels there is a good correlation between the subjective similarity and the rates of false identifications, but not for sentences. Speech transmitted by usual telephone handsets, applied with an acoustical leakage to the ear, results in very high false identification rates (up to 45%).

5aSP9. Acoustic features as indices of emotional responding following reward and failure feedback. Jo-Anne Bachorowski, Christopher D. Linker, and Michael J. Owren (Dept. of Psychol., Univ. of Colorado, Campus Box 173, P. O. Box 173364, Denver, CO 80217-3364)

Comprehensive models of personality regard individual differences in emotional responding as important mediators of behavior across a variety of contexts. However, traditional psychophysiological and self-report measures are often unreliable or unwieldy when applied to emotional responses. Several acoustic features were examined as indices of changes in emotional state. Adult subjects were given noncontingent positive and negative feedback as they participated in a computerized lexical decision task. Acoustic components were extracted from discrete

speech segments recorded immediately following feedback. In comparison to baseline recordings, higher $F0$ values were associated with both positive and negative feedback. However, changes in jitter and shimmer were differentially associated with the type of feedback in that increases in these perturbation measures were especially apparent following negative feedback. The results of this research will be used to elaborate the cognitive and emotional processing of trait-anxious and trait-impulsive individuals in contexts designed to engender anxious and impulsive behavioral responses.

5aSP10. Perception of multiple levels of prominence in spontaneous speech. Colin W. Wightman (Dept. of Elec. Eng., New Mexico Inst. of Mining and Technol., Socorro, NM 87801)

In both read and spontaneous speech, phrasal prominences play an important role in conveying the speaker's intent. Prominences serve both to mark important discourse-related events in the conversation and help to resolve ambiguities at several levels. Many speech researchers report the intuition that there are several levels of prominence, that is, that some prominences are bigger than others. Nonetheless, attempts to train human labelers to mark multiple levels of prominence have not been successful: while there was agreement on the location of prominences, there was little agreement between labelers on the level to be assigned to each. Here, an alternative approach, using a panel of naive listeners to mark prominences in a corpus of spontaneous speech has been taken. Instead of marking multiple levels of prominence, a simple binary labeling was used by each labeler and the level of each prominence determined by the number of labelers marking it. In this paper, the results of this preliminary study are presented, and the relationships between the estimated levels of prominence and the acoustic correlates of vowel lengthening, syllable lengthening, and pitch level and excursion are investigated.

5aSP11. Inferring articulatory movements from acoustic properties at fricative-vowel boundaries. Lorin Wilde (Res. Lab. of Electron., Dept. of Elec. Eng. and Comput. Sci., MIT, Rm. 36-547, Cambridge, MA 02139)

Formant transitions provide context-dependent acoustic cues that can be interpreted in terms of the articulatory kinematics involved in moving from a consonant to a vowel. Formant frequencies were measured at identified acoustic landmarks for eight English fricatives preceding front, back, and back-rounded vowels. Formant onset times designated the point when the energy increased most rapidly and evidence of the first formant was first observed. Comparing the two-dimensional representation of $F2 \times F3$ onset frequencies along the voicing dimension showed the voiceless fricatives to be more dependent on vowel context. The onset frequencies for voiced fricatives reflect a more extreme supraglottal posture, while the voiceless fricative measures can be considered to be at a point closer to the vowel because voicing begins at a later time relative to the oral release gesture. Formant structure in the noise before the release, to the extent that it is visible in the consonantal interval prior to voicing onset, can provide additional place information for voiceless fricatives. Formant onset data are compared with measurements reported from investigations of place categorization of stop consonants. [Work supported by NIH.]

5aSP12. Can you rustle up some attention?: Further results on breath sounds in synthesis. Sonya M. Sheffert (Haskins Labs., 270 Crown St., New Haven, CT 06511, and Univ. of Connecticut), D. H. Whalen (Haskins Labs., New Haven, CT 06511), and Charles E. Hoequist (BNR, Inc., Research Triangle Park, NC 27709)

During conversation, speakers typically take a breath when preparing to speak. At the previous meeting [Whalen and Hoequist, *J. Acoust. Soc. Am.* **93**, 2298(A) (1993)], it was shown that subjects were significantly more accurate at transcribing synthesized sentences that were preceded by a (naturally produced) breath intake than those that were not. It was concluded that the perception of synthetic speech is facilitated by the addition of a breath, presumably because it increased the naturalness of the stimuli. However, there is the possibility that this result is simply due to the extra attention-getting effect of the breath between the warning tone and the sentence. To test this possibility, the

breath was replaced with the sound of rustling leaves. The rustling sounds matched the breaths in duration and loudness. The results should tell us whether the presence or absence of a breath before a synthesized sentence functions as a cue for the upcoming speech, or merely as a general attention-orienting sound. [Work supported by NIH Grant No. HD-01994.]

5aSP13. The preliminary application of Gabor spectrogram analysis in speech samples. Jack J. Jiang (Dept. of Otolaryngology—Head and Neck Surgery, Northwestern Univ., 303 E. Chicago Ave., Chicago, IL 60611), Shie Qian (Natl. Instruments, TX 78730), David G. Hanson, Eric Cuasay, and Jerilyn Logemann (Northwestern Univ., Chicago, IL 60611)

It has been widely recognized that the FFT-based spectrogram does not provide good simultaneous resolution in both time and frequency domains. A new method of spectral analysis has been developed based upon the Gabor expansion and the Wigner-Ville distribution. The resolution of the Gabor spectrogram is twice as high as that of a FFT-based spectrogram. In this report, FFT-based spectrograms and Gabor spectrograms are compared for 5 English vowels, 6 stops consonants, 4 fricatives, and vowels format transitions in a CVD contents on 6 normal subjects. Results demonstrate that the Gabor spectrogram is a promising alternative for FFT-based spectrogram in speech analysis because of its higher temporal and frequency resolution.

5aSP14. Searching for attractors in speech and nonspeech respiration. Robert J. Porter (Kresge Hear. Res. Lab. of the South, Dept. of Otorhinolaryngol., Louisiana State Univ. Med. Ctr., New Orleans, LA 70112) and David M. Hogue (Univ. of New Orleans, New Orleans, LA 70148)

Our recent study of breathing during speech revealed that a nonlinear equation, originally proposed as a description of respiration during quiet breathing, may also provide a description of respiratory behavior during speech. This equation interrelates variables such as lung volume, airway resistance, and the duration of the utterance, and captures commonalities in the basic organization of the respiratory system during what have been considered different tasks or modes of behavior (i.e., speech and nonspeech). Further explorations of these data suggest that the mechanism underlying the behavior may be viewed as an attractor, possibly chaotic. The presence of such an attractor in respiratory patterns is supported by the findings of other researchers and is consistent with the adaptive nature of the system during speech and nonspeech tasks [e.g., M. P. Sammon and E. N. Bruce, *J. Appl. Physiol.* **70**(4), 1748–1762 (1991)]. Descriptions of the attractor and some of its characteristics will be presented.

5aSP15. Velar coarticulation revisited. Fredericka Bell-Berti (Dept. of Speech, Commun. Sci., and Theatre, St. John's Univ., Jamaica, NY 11439 and Haskins Labs., 270 Crown St., New Haven, CT 06511), Rena A. Krakow (Temple Univ., Philadelphia, PA 19122 and Haskins Labs., New Haven, CT 06511), and Dorothy Ross (CUNY Graduate School, New York, NY 10036 and Haskins Labs., New Haven, CT 06511)

Several models, embodying fundamentally different assumptions about the nature and organization of speech motor control, have been offered in explanation of observations of coarticulatory phenomena. A series of experiments has been undertaken to test specific predictions about the role of carryover coarticulation in delaying the normally stable onset of a velar gesture for a segment, made by Bell-Berti and Harris' coproduction model [F. Bell-Berti and K. S. Harris, *Phonetica* **38**, 9–20 (1981); F. Bell-Berti, *Phonet. Phonol.* **5**, 63–85 (1993)]. For example, the onset of velar lowering for a vowel or a nasal consonant following an obstruent will be delayed until near the end of the acoustic period of the obstruent (depending upon its duration), to insure adequate velopharyngeal closure for the obstruent. (That is, the velar gesture for a later segment is modified by the requirements for a preceding one.) Early results encourage the belief that it will be possible to account for anticipatory and carryover effects with a single model. [Work supported by NIDCD Grant No. DC-00121 to the Haskins Laboratories.]

5aSP16. The effect of learning due to voice assessment in acoustic analysis of vocal tremor in patients with Parkinson's disease. Judith B. King (Recording and Res. Ctr., Denver Ctr. for the Performing Arts, 1245 Champa St., Denver, CO 80204 and Northern Arizona Univ., College of Health Professions, Box 15045, Flagstaff, AZ 86011), Jon H. Lemke (Univ. of Iowa, Iowa City, IA 52242), and William S. Winholtz (Denver Ctr. for the Performing Arts, Denver, CO 80204)

For years researchers have questioned the influence of learning from baseline assessments. In a 3-yr study of 14 Parkinson's patients who received no speech treatment, investigators studied phonatory variability and the effect of learning due to voice assessment. Statistically significant constant decline in weighted mean values were found to exist over time on acoustic variables of maximum performance. A strong learning experience as a result of baseline assessment was also documented. However, measures of maximum performance are notoriously variable, potentially unstable and have been linked to effort, motivation, and learning. Therefore, the present study was designed to examine the effect of learning reflected among the same group of 14 subjects but on a more sensitive, less motivationally dependent phonatory variable. Amplitude- and frequency-demodulated outputs and measures of frequency and level (percent) of low-frequency vocal tremor components in sustained phonation were analyzed. Growth curve analysis models were constructed to simultaneously estimate a constant rate of decline of the variables over time (excluding the influence of baseline values) and to estimate the effect of learning due to baseline assessments. Preliminary data from this study enhance the understanding of the progressive nature of Parkinson's disease among untreated subjects, and further define which acoustic measures of voice are more sensitive to the effect of learning. [Work supported in part by DC 00976 from the National Institute on Deafness and Other Communication Disorders; Organized Research Grant, Northern Arizona University.]

5aSP17. Changes in voice-onset time in speakers with cochlear implants. Harlan Lane, Jane Wozniak, Joseph Perkell, Melanie Matthies, and Mario Svirsky (Res. Lab. of Electron., Rm. 36-511, MIT, Cambridge, MA 02139)

Voice-onset time (VOT) was measured for the English plosives in /Co/ context spoken by three female and one male postlingually deafened recipients of multi-channel (Ineraid) cochlear implants. Recordings were made of their speech before activation of their speech processors and at intervals after activation, extending over several years. Also measured were: instantaneous oral airflow; sound pressure level; an indirect index of glottal aperture; and average airflow during passage readings. Pre-implant, all four speakers characteristically uttered plosives with too-short VOT, compared to normative data. After activation of their processors, all four were relatively accurate in identifying plosives with respect to voicing, and the three female speakers lengthened VOT. The women also increased glottal aperture post-activation. However, none of the women increased peak oral flow following plosive release, possibly because of a countervailing decrease in subglottal pressure: all three reduced SPL post-activation. Complementary results were found for the male speaker: a decrease in inferred glottal aperture and an increase in oral flow accompanied by an increase in SPL. Increases in glottal aperture are expected to inhibit the onset of voicing in the plosives [K. Stevens, *Phonetica* 34, 264-279 (1977)]. Consequently, some or all of the observed increases in VOT with activation of the implant processors may be due to "postural" adjustments of the larynx. [Work supported by N.I.D.C.D.]

5aSP18. Release-from-masking effects provided by a hearing aid digital signal processor. Michael A. Grim (Dept. of C.D.S.S., Univ. of Colorado, Campus Box 409, Boulder, CO 80309), Christopher Schwietzer, Eric Lindemann (AudioLogic, Inc., Gunbarrel, CO 80027), and Richard H. Sweetman (Univ. of Colorado, Boulder, CO 80309)

Release-from-masking effects provided by a digital hearing aid signal processor utilizing multiple-microphone inputs were evaluated with three measures of speech recognition. Speech recognition measures were monosyllabic word recognition score, reaction time, and subjective rat-

ing of the intelligibility of selected passages of continuous discourse. Microphones, placed on KEMAR, recorded speech stimuli embedded in cafeteria noise yielding S/N ratios of 0 and 8 dB SPL. Half of the speech-in-noise stimuli were processed through the hearing aid digital signal processor while half remained unprocessed. The hearing aid processor utilizes a technology similar to adaptive-beamforming to reduce the masking effects of background noise on speech recognition. Unprocessed and processed speech-in-noise stimuli were presented to 10 normally hearing subjects, 10 hearing-impaired subjects, and 10 hearing-impaired individuals fit with linear amplification. Comparison of word recognition scores, reaction times, and intelligibility ratings for the two S/N ratios between unprocessed and processed speech-in-noise stimuli suggest that the hearing aid processing scheme provides significant release-from-masking effects which may improve the recognition of speech in noise for normally hearing and hearing-impaired listeners.

5aSP19. Dishabituation of visual attention by natural and synthetic infant-directed speech stimuli varying in spectral properties. Peter S. Kaplan, Michael H. Goldstein, and Michael J. Owren (Psychol. Dept., Univ. of Colorado, P.O. Box 173364, Denver, CO 80217-3364)

A habituation-dishabituation paradigm was used to study acoustic characteristics of adults' infant-directed (ID) speech that elicit infant visual attention. Five groups of 4-month-olds received twelve 10-s presentations of a checkerboard pattern. On the ninth trial, one of five auditory stimuli was presented. Differences in looking duration were compared on the trials before and after the auditory stimulus. Attention increased significantly after presentations of a natural, intact ID speech segment and after a version composed of sine waves simulating its fundamental frequency (F_0) and first five harmonics (H_1-H_5). However, no significant changes in attention occurred following sine waves simulating the F_0 only, the F_0 -plus- H_1 , or the harmonics only. The harmonics-only stimulus elicited significantly less responding than did any other stimulus. The roles of frequency modulation and spectral properties are discussed.

5aSP20. Perceptual evaluation of infant articulatory transitions. Nancye C. Roussel (Dept. of Commun. Disord., Univ. of Southwestern Louisiana, P. O. Box 43170 USL, Lafayette, LA 70504) and Raymond G. Daniloff (North Texas State Univ., Denton, TX)

The distribution of phonemic cues contained within consonant-vowel (CV) nonreduplicated (NRB) syllables of infants ages 6 to 15 months was compared to the distribution observed in adult productions [S. Furui, *J. Acoust. Soc. Am.* 57, 1016-1025 (1986)] to examine the extent of coproductive overlap of infant consonant and vowel gestures. Truncated versions of infant CV syllables were presented to adult listeners for syllable identification. Perceptual critical points (truncation point at which syllable identification fell below 60% for the first time) were determined for both initial and final truncation conditions, and critical intervals were calculated. Results revealed significant differences in the temporal distribution and strength of phonemic cues within the infant NRB CV syllable. Fewer than 1/3 of the infant syllables demonstrated normal, adult-like identification functions under both initial and final truncation conditions, and infant critical intervals were on the average $2\frac{1}{2}$ times longer than those of adult Japanese speakers. Results did not reveal a clear developmental progression towards either more adult-like critical interval durations or identification functions in the NRB syllables studied.

5aSP21. Intrinsic F_0 in the babbling of infants at ages 6, 9, and 12 months. D. H. Whalen (Haskins Labs., 270 Crown St., New Haven, CT 06511), Andrea G. Levitt (Haskins Labs. and Wellesley College), Iris Smorodinsky (Haskins Labs. and Yale Univ.), and Pai-Ling Hsiao (Haskins Labs. and Univ. of Connecticut)

In adult speech, fundamental frequency (F_0) has been found to be relatively high with high vowels and low with low vowels. This "intrinsic pitch" has been found in every language examined, including tone languages. One study with infants [Bauer, *Folia Phonia*. 40, 138-146 (1988)] found that F_0 changed consistently with the front/back dimension rather than the high/low dimension. Bauer interpreted this pattern in terms of the change in vocal tract shape that occurs over the first 18

months of life. In this study, four infants were examined at 6, 9, and 12 months of age. Only vowels with non-mid-central articulation were selected. Results indicated that there was, after all, an association with height, though there was also an interaction with front/back. Reasons for the discrepancy with earlier work will be discussed. [Work supported by NIH Grant No. DC-00403.]

5aSP22. Children's control of temporal variability in repetition tasks. Bruce L. Smith and Mary Kay Kenney (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2299 Campus Dr. N, Evanston, IL 60208)

Although several previous studies of normally developing children have examined effects of speaking rate changes on segment duration and variability and their possible relationship to neuromotoric maturity for speech, there have been no investigations concerned with children's ability to directly control the variability of their speech. The present study attempted to evaluate children's control of temporal variability by assessing the performance of 7- to 11-year-old subjects who were asked to minimize variability in their speech. The tasks included: (1) a control condition, in which 11 subjects repeated a variety of stimuli 25 times each, with no special instructions as to how they were supposed to produce them, and (2) an experimental task, in which subjects were specifically instructed to be as consistent as possible in repeating stimuli in an attempt to minimize variability. Results indicate that for the group as a whole, the temporal variability of various speech segments did not tend to differ substantially when comparing the children's control and experimental productions, with approximately half the subjects showing small decreases in variability and half showing small increases for the segments that were measured. It, thus, appears that for at least this type of repetition task, subjects perform at essentially optimal levels, in terms of variability, even when not specifically attempting to do so.

5aSP23. Syllable duration in Italian and Japanese. Laura Brighenti (Inst. for Speech and Language Sci., New York Univ., New York, NY 10003) and Peter Homel (Baruch College, CUNY, New York, NY 10003)

This study analyzes duration of stop syllables in Italian and Japanese. Its results show that the two languages are very similar in terms of syllable length. They also indicate that, in accordance with previous findings, both Italian and Japanese syllable duration is contrastive in terms of gemination, and Italian duration is also contrastive as a function of stress. Finally, they show that both languages present variability in the duration of their syllables, even in cases where the syllable is neither geminate nor stressed. These results point to the following conclusions: (1) other elements outside the syllable itself (i.e., position within word, surrounding syllables, information content, etc.) may be involved in determining syllable duration; and (2) the stress-timed/syllable-timed language distinction may not be useful for understanding how languages assign duration for particular parts of the syllable. In particular, the results of this study indicate that stress and syllable position play an important role in determining syllable duration even in syllable-timed languages.

5aSP24. Perception of stop consonants in speech signals reconstructed from phase or amplitude. L. Liu, J. L. He, and G. Palm (Dept. Neuroinformation, Univ. of Ulm, 7900 Ulm, Germany)

It is well known that under certain conditions, a signal is completely specified either by its Fourier transform phase or amplitude. An iterative technique similar to that used by Hayes *et al.* [IEEE Trans. Acoust., Speech, Signal Process. ASSP-28, 672-680 (1980)] was applied to the global Fourier transform phases or amplitudes of natural VCV (vowel-stop consonant-vowel) utterances and the perception of stop consonants was experimentally studied. Under a variety of conditions, the consonant identification performance in the phase-only stimuli improved from about 66% to more than 90% after the iterations. The influence of the initial amplitude guess on the quality of the reconstructed signals was also investigated. The retrieval of stop consonant information from phase for various window lengths and various phase noise levels were analyzed with regard to vowel contexts, stop manner, and place of articulation. In contrast to the case of signal reconstruction

from its phase, the stimulus reconstructed from the amplitude was found not of much value in representing the original VCV signal. Little improvement on consonant perception could be made after the iterations if a flat or a random phase was taken as the initial guess. However, with an appropriate choice of the initial guess, which contains one bit of the original phase information, stimuli with near-perfect consonant identification performance were reconstructed.

5aSP25. A multidimensional scaling analysis of vowel discrimination in humans and monkeys. Joan M. Sinnott, Charles H. Brown, and Regina A. Kressley (Comparative Hear. Lab., Dept. of Psychol., Univ. of South Alabama, Mobile, AL 36688)

Five humans and three African Sykes' monkeys (*Cercopithecus albogularis*) discriminated among the ten English vowels using a repeating background procedure. Human subjects are four native American-English speakers and one non-native Hispanic speaker. Reaction times were input to a multidimensional scaling analysis (ALSCAL) in order to derive a measure of perceived similarity or dissimilarity among the vowels. For all subjects, including monkeys, the front vowels were the most distinguishable vowel group, while the central and back vowels were less clearly differentiated. The Hispanic speaker performed similarly to the native American English speakers, although Spanish does not differentiate among the spectrally similar vowels of English. One difference that emerged between humans and monkeys was that humans appeared more sensitive than monkeys to first formant changes in the front vowels. [Work supported by NIDCD.]

5aSP26. Can lexical knowledge inhibit phoneme perception? Lee H. Wurm and Arthur G. Samuel (Dept. of Psychol., State Univ. of New York, Stony Brook, NY 11794)

This series of experiments tests a critical prediction of the TRACE model of speech perception [J. L. McClelland and J. L. Elman, Cog. Psychol. 18, 1-86 (1986)]. Experiment 1 was a replication [U. H. Frauenfelder, J. Segui, and T. Dijkstra, JEP:HPP 16, 77-91 (1990)]. Subjects listened to lists of words and non-words, and made speeded detection responses to specified phoneme targets. In accord with the Frauenfelder *et al.* result, experiment 1 produced no evidence of inhibitory lexical effects on phoneme monitoring reaction times. Experiment 2 improved several design aspects of the Frauenfelder *et al.* experiment by using balanced target locations, increasing the number of critical stimuli, and using a more appropriate baseline condition against which to measure inhibitory effects. The results of this line of research clarify whether indirect lexical inhibition should be included in models of lexical access.

5aSP27. Lexical effects in nonwords? Rochelle Newman, James R. Sawusch, and Paul Luce (Dept. of Psychol., SUNY, Buffalo, NY 14260)

Previous research has reported the existence of a "lexical effect" in identification tasks. For instance, one series might range from "dice" to "tice," while the other varied from "dype" to "type." The ambiguous stimuli from each series were more likely to be classified as members of the category which makes a word. (Thus, they would be classified as "d" in the dice-tice series but as "t" in the dype-type series.) However, these results have sometimes been inconsistent, suggesting that factors other than lexical status are involved. In a series of experiments, pairs of nonword-nonword continua were presented to listeners, where one end of each series had a higher neighborhood density than the other end. That is, one end of each series was similar to a greater number of words, weighted according to frequency of occurrence, than the other. Ambiguous items were classified by listeners with the phonetic label corresponding to the higher density end of the series. The results suggest that neighborhood effects may contribute to the variability in previous "lexical" effects. [Supported by NIDCD Grant No. DC00219 to SUNY at Buffalo and an NSF Graduate Fellowship to the first author.]

5aSP28. Transformed auditory feedback: Effects of fundamental frequency perturbation. Hideki Kawahara (ATR Human Information Processing Res. Labs., 2-2 Hikaridai Seika-cho, Soraku-gun, Kyoto, 619-02 Japan)

A rapid interaction between auditory perception and speech production has been found using the technique of transformed auditory feedback with fundamental frequency perturbation as the transformation. The technique of transformed auditory feedback has been developed to investigate the role of auditory perception in speech production under natural conditions. The basic idea of the method is to keep the disturbance of normal speech production processes to a minimum, while allowing interactions between speech production and auditory perception to be detectable. In the first experiment, subjects were instructed to sustain the Japanese vowel /a/ with a constant pitch. The phonated vowel was frequency modulated using sinusoids of 2- to 7-Hz range and feedback diotically via headphones. The modulation depth was 200 cent from peak to peak. The results indicate that there is a phase-locking effect in the fundamental frequency of the produced speech. In a second experiment, a correlation analysis using a pseudo-random signal as a modulator revealed that the reaction to fundamental frequency perturbation is corrective and that its latency ranges from 100 to 200 ms. Its relation with the auditory-laryngeal reflex will be discussed.

5aSP29. The perception of chirps by Sykes' monkeys and humans. Charles H. Brown, Joan M. Sinnott, and Regina A. Kressley (Dept. of

Psychol., Univ. of South Alabama, Mobile, AL 36688)

Four Sykes' monkeys (*Cercopithecus albogularis*) and three humans discriminated among 12 chirps presented in a repeating background paradigm. The test stimuli consisted of sets of four chirps recorded from Sykes' monkeys, red-tailed monkeys (*Ce. ascanius*), and small East African birds, respectively. Reaction times were submitted to a multi-dimensional scaling analysis (ALSCAL). All monkey listeners perceived the bird chirps as similar to each other, and distinct from the monkey calls, while two of the three human listeners had difficulty distinguishing the bird chirps from the monkey calls. Both human and monkey subjects tended to exhibit overlapping maps for the Sykes' and red-tailed calls, but the magnitude of overlap tended to be greatest for the monkey listeners. Guenon monkeys give chirps in similar alarm contexts, and these calls alert members of sympatric species to common dangers. Conversely, bird chirps are very commonly heard, are not associated with alarm, and must constitute for monkeys an acoustical impediment to communication. The present data suggest that the monkey's perceptual map of these highly similar calls is influenced by the biological significance of these calls in nature. [Work supported by NIDCD.]

FRIDAY MORNING, 8 OCTOBER 1993

COLUMBINE ROOM, 7:55 A.M. TO 12:00 NOON

Session 5aUW

Underwater Acoustics: Propagation

Kevin L. Williams, Chair

Applied Physics Laboratory, University of Washington, Seattle, Washington 98105

Chair's Introduction—7:55

Contributed Papers

8:00

5aUW1. Implementation of generalized impedance boundary condition in the split-step Fourier parabolic equation. F. J. Ryan (Ocean and Atmospheric Sci. Div., Code 541, NReD, San Diego, CA 92152-5000)

The parabolic wave equation (PE) is a powerful numerical method for computing the full-wave complex acoustic pressure field in range-dependent environments. The split-step Fourier PE algorithm (SSFPE) of Hardin and Tappert provides a very efficient computational implementation of PE when the surface boundary condition is of the Dirichlet or Neumann form, and when the solution decays sufficiently with depth. This allows use of fast Fourier transform (FFT) methods to implement the SSFPE algorithm. Typically the infinite domain radiation condition is approximated on a finite calculation grid by employing an artificial absorber or sponge. In some cases involving strong bottom interaction, however, the required absorber region may be quite large. This results in an increased computational load. A new method for truncating the absorber region will be described that is based upon splitting the PE field into two components that are in turn propagated on different vertical wave-number grids. Illustrative examples that use this new method will be shown. [Work supported by NReD IR program.]

8:15

5aUW2. Minimax rational approximations for the parabolic equation method. Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Higher-order Padé approximations have been applied to reduce the limitations and to improve the efficiency of the parabolic equation (PE)

method [M. D. Collins, *J. Acoust. Soc. Am.* **93**, 1736-1742 (1993)]. Several of the derivatives are correct at the origin for Padé approximations. Improved efficiency may be achieved by using rational approximations designed using the minimax method, which involves minimizing the maximum error over a domain of interest. This approach has been applied for deriving a PE for fluid media that handles wide propagation angles [Vefring and Mjølness, *J. Acoust. Soc. Am.* **93**, 1736-1742 (1993)]. Additional applications of the minimax approach include deriving improved approximations for the elastic PE (for which stability is an important issue), the two-way PE (which may require a careful treatment of the evanescent spectrum), and the split-step Padé solution (which involves that approximation of a relatively complicated function).

8:30

5aUW3. Comparison of a parabolic equation model with Gaussian beam tracing in range-independent and range-dependent environments. Timothy H. Ruppel and Robert L. Field (Naval Res. Lab., Stennis Space Center, MS 39529)

Transmission losses predicted by the parabolic equation model FEPE over a 20- to 150-Hz frequency band are compared with transmission losses predicted by Gaussian beam tracing in range-independent and downslope environments. Ocean impulse responses, computed with each model, are also compared with experimentally determined responses previously reported. [R. L. Field and J. H. Leclere, *J. Acoust. Soc. Am.* **93**, 2599-2616 (1993)]. As in the earlier paper, environmental inputs to the model are derived from the Ocean Drilling Project (ODP) sites near the experiment location, and impulse response comparisons are made using correlation coefficients. [Work supported by the Office of

8:45

5aUW4. Modeling acoustic particle velocity in range-dependent environments with a parabolic equation code. K. B. Smith, G. L. D'Spain, and W. S. Hodgkiss (Marine Phys. Lab., Scripps Inst. of Oceanogr., San Diego, CA 92152-6400)

Measurements have been made of the acoustic particle velocity field generated by a towed source transmitting 7-, 10-, and 16-Hz tones. The periodic change in the polarization of the motion as a function of source-receiver range, from prograde circular motion to retrograde circular motion and back, appears to have been disrupted by changes in slope of the bottom topographic features in the experiment. In order to test this hypothesis, the University of Miami parabolic equation code (UMPE) was modified in order to efficiently calculate acoustic particle velocity. This numerical code makes effective use of the equivalent fluid approximation for low shear speed ocean bottoms [C. T. Tindle and Z. Y. Zhang, *J. Acoust. Soc. Am.* **91**, 3248-3256 (1992)]. The predictions of the UMPE model are compared with the actual measurements, and with the predictions made by the finite element parabolic equation (FEPE) by M. D. Collins, which explicitly incorporates the effects of bottom shear. [Work supported by ONR.]

9:00

5aUW5. Long-range acoustic propagation in four dimensions. Gregory J. Orris and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Ocean acoustic modeling is usually restricted to frequency-domain problems in two dimensions. Since ocean acoustics problems typically involve domains that are very large relative to a wavelength, computation times often become prohibitive when a third dimension (either time or the third spatial variable) is included. The adiabatic mode parabolic equation (PE), which has been applied to solve global-scale problems in three spatial dimensions [Collins *et al.*, *J. Acoust. Soc. Am.* **93**, 2321 (1993)], is practical for solving four-dimensional problems out to long range (i.e., hundreds of wavelengths or more). In the time domain, the adiabatic mode PE solution illustrates the flow of energy in the modal coefficients, which depend on range and azimuth.

9:15

5aUW6. Propagation and signal coherence in bottom-limited shallow water environments. F. J. Ryan (Ocean and Atmospheric Sci. Div., Code 541, NRaD, San Diego, CA 92152-5000) and L. A. Shook (NRaD, San Diego, CA 92152-5000)

Simulation results will be presented for complex shallow water environments off the southern California coast. The study focuses on down-slope propagation from sources in shallow water to deeper bottomed sensors. The environment is a thinly sedimented series of terraces separated by steep slopes. A very high fidelity geoacoustic bottom model, SEABEAM-type bathymetry and dense synoptic oceanographic measurements are used to compute pressure fields using PE codes. Examples of spatial signal coherence and spatial variability will be shown. The question of the effects of environmental data spatial resolution (bathymetry and sub-bottom) on computed signal fields will be addressed.

9:30

5aUW7. A fast one-way normal mode propagation method. E. White and M. F. Werby (NRL, Theoretical Acoustics, Code 7181, Stennis Space Center, MS 39529)

The development of a new fast normal mode method produces vertical wave functions that are in closed form. One-way coupled mode methods can be based on projection matrices that are generated by integral overlaps between vertical wave function of adjacent domains.

For methods that require numerical solutions of the vertical wave functions the computation of the matrix elements is very time consuming and can even lead to inaccuracies at higher frequencies. However the closed form solutions allow one to code matrix elements in relatively simple closed form based on simple arithmetic sums. This leads to a rapid method for generating the projection matrices and therefore connects the stepwise domains in a rapid fashion. The method is tested and compared with other available coupled mode programs. Ultimately the method will be used for the object in a waveguide program for which this method was originally designed. [Work supported by Naval Research Laboratory 6.1 program (element 601153N) with technical management provided by NRL-SSC.]

9:45

5aUW8. New algorithm for calculating normal modes in broadband applications. Robert A. Koch and D. P. Knobles (Appl. Res. Lab., Univ. of Texas, P. O. Box 8029, Austin, TX 78713-8029)

A new algorithm for calculating the normal modes that are used to describe underwater acoustic propagation is presented. The algorithm is especially relevant for broadband normal mode calculations in which normal mode eigenvalues, depth functions, and attenuations are needed. The new algorithm uses an expansion technique to construct normal mode eigenvalues and depth functions for some frequency from eigenvalues and depth functions previously calculated at another, higher frequency. Comparisons of the estimated eigenvalues, depth functions, and attenuations with exact results are presented for several shallow water environments. A broadband propagation model that uses normal mode sets calculated with the new algorithm has been developed. Transmission loss and impulse responses based on estimated and exact normal mode sets are compared. [This work was supported by the Office of Naval Research through Block RL3B and Block CS3A.]

10:00-10:15 Break

10:15

5aUW9. Three-dimensional modeling of acoustic pulse propagation through shallow water. Robert T. Chapman and Donald J. McDowell, Jr. (Naval Undersea Warfare Ctr., Div. Newport, Detachment New London, Code 423, New London, CT 06320)

A three-dimensional model capable of simulating the propagation, scattering, and reverberation of acoustic waveforms with an arbitrary time dependence is presented. The finite element method is used to model the spatial characteristics of the shallow water domain. An explicit integration scheme is used to obtain the time-dependent pressure field solution, for the entire computational space, due to a prescribed source function. Multiple independent sources, depth- and range-dependent bathymetry, volumetric scattering, fluid sediments, and scattering from irregular bottom topography may be considered with this modeling technique. As implemented, the computational scheme utilizes enhanced supercomputer vector processing, and an element-by-element storage strategy to enhance the computational efficiency and feasibility for the three-dimensional, time-dependent problems of interest. Computational results will be presented for the propagation of a finite duration pulse from a point source in deep water, up a sloping bottom, into shallow water. [Work supported by the NUWC IR and IED programs.]

10:30

5aUW10. Factorization of propagator matrices for efficient computation of wave-fields in horizontally layered fluid-solid media. Sven Ivansson [Dept. of Hydroacoust. and Seismol., Natl. Defence Res. Establishment (FOA 260), S-172 90 Sundbyberg, Sweden]

The wave field is decomposed into its frequency-wave-number components. Compound matrices for solid layers provide a convenient way of computing the boundary values at a fluid-solid interface [M. B. Porter and E. L. Reiss, *J. Acoust. Soc. Am.* **77**, 1760-1767 (1985)], with

loss-of-precision control. A certain vector is propagated through a sequence of multiplications with compound matrices, one for each layer. It is shown that computations of this kind can be performed more efficiently if each compound matrix is decomposed as a product of *sparse* matrices that are applied in sequence. Two kinds of compound-matrix factorizations are proposed. In connection with dispersion computations, our first factorization gives a method that is related to the "fast form" of Knopoff's method [F. Schwab *et al.*, Bull. Seismol. Soc. Am. 74, 1555-1578 (1984)]. This algorithm is slightly more efficient, however, and its range of applicability is wider. The second compound-matrix factorization gives a method that is *significantly* faster than the "fast form" of Knopoff's method. Very few arithmetic operations are needed. It also provides a good basis for analyzing the numerical performance of compound-matrix propagation. Finally, it is shown how propagator-matrix factorization can be used to enhance the efficiency for multi-frequency computations and computation of *full* wave-fields, by wave-number integration or modal synthesis.

10:45

5aUW11. Analysis of the acoustic field via spectral decomposition. Guy V. Norton (NRL-SSC, Stennis Space Center, MS 39529-5004)

Spectral decomposition has been used by various authors as a calculation technique for acoustic propagation in range independent and recently range dependent ocean environments. One of the methods' strong points is that it maps the initial acoustic field at each given depth point in the waveguide into horizontal wavenumber space. Since the field is now expressed in wavenumber space, analogous to normal mode theory, various acoustic quantities can easily be calculated. These quantities such as the group velocity or the incident field expressed in terms of the plane wave spectrum in the past were calculated using normal modes. However, since all that is required to initiate the method is a complex acoustic field evenly space in depth of the waveguide, the method has been coupled to the output of a PE model, in order to analyze that field. The associated theory and numerical examples will be given. [Work sponsored by ONR.]

11:00

5aUW12. Frequency-dependent sound attenuation in the northeastern Pacific Ocean below 400 Hz. Gerald B. Morris (Code 7172, Naval Res. Lab., Stennis Space Center, MS 39529)

Signals from SUS explosives were received at four hydrophones suspended in the deep ocean from the Research Platform FLIP. Energy flux densities were determined for approximately 700 of these broadband sources at ranges from 70 to 1250 nmi. The normal approach in determining attenuation is to calculate the transmission loss with no frequency-dependent attenuation as a function of range using a numerical model, subtract the calculated spreading loss and attribute the remaining loss to frequency-dependent attenuation. Because of the large losses due to geometric spreading and small losses from the attenuation together with the strongly range-dependent environment calculating the spreading or transmission loss with high accuracy is difficult. An alternate technique was used. Differences in measured propagation losses at various frequency bands up to 400 Hz were used to calculate relative losses referenced to a band near 50 Hz. Assumptions were then made regarding the functional form of the frequency dependency of these losses and the resulting frequency-dependent attenuation losses determined. The resulting values are consistent with a boric acid relaxation frequency of 1 kHz, but the loss coefficient is lower than those predicted from Thorp's equation. These results are in agreement with published values determined for higher frequencies from cw acoustic transmissions in the same region. [Work supported by ONR.]

11:15

5aUW13. Some observations of acoustic variability and array performance at short range near Vestfjorden. Donald R. Del Balzo (SACLANT Undersea Res. Ctr., Viale San Bartolomeo, 400, 19138 La Spezia, Italy)

An experiment was conducted in October 1989 in shallow water (250-320 m) near Vestfjorden in support of sonar system research at SACLANTCEN. This talk describes some observations of acoustic variability and array performance at short range for one-way pulse propagation during a 46-min period. The signals were 0.5-s LFM pulses in the frequency band of 340-345 Hz. They were received at a range of 13 km on 59 elements of a towed array. Large variability in the received signal level across the array, by as much as 10-15 dB, was observed on some individual pulses. These acoustic variations appear to be related to depth variability along the array due to a small array tilt. Normal mode calculations in this environment indicate the existence of a deep acoustic null in the vicinity of the measured data. Average array signal gain degradation for all data with SNR > 20 dB is only 0.6 dB, so the array beamforming performance is close to the theoretical maximum on this 27-wavelength aperture in shallow water.

11:30

5aUW14. Shell function method incorporated ray representation for computing the acoustic pressure field in inhomogeneous media. Haitao Pan and Shigeo Ohstuki (Precision and Intelligence Lab., Tokyo Inst. of Technol., Nagatsuta 4259, Midori-ku, Yokohama, 227 Japan)

The Shell function method is presented as a new technique to evaluate the acoustic pressure field, produced by a practical source of finite area, in a medium having an arbitrary spatially varying sound speed. The Shell function method considers the acoustic wave received at the observation point, for a specified travel time, and the associated vibrating source area. In terms of ray representation, this approach is accomplished by interchanging the position of the observation point with that of the elemental area of the source. Both the convergent state of a sound beam issuing from the observation point, and the element area of the source responsible to this convergent state, are taken into account simultaneously. As a consequence, the sound pressure at, and in the vicinity of, caustics can be given an appropriate value directly. The Shell function method was evaluated by the computation of the acoustic field in a depth-dependent sound-speed profile containing caustics, for a linear source with a specified length.

11:45

5aUW15. The calculation of time domain signals in a waveguide with applications. M. F. Werby and E. White (NRL, Theoretical Acoustics, Code 7181, Stennis Space Center, MS 39529)

A fast normal mode code has been developed that is based on an expansion method and a fast Fourier transform integral method. This has enabled one to perform broadband pulse calculations to high frequencies quickly. Further the pulse returns can be decomposed into their constituent modes. Studies on pulse returns were performed for realistic shallow water waveguides for signals out to 25 km and the various dispersion effects and separate model arrivals with increasing distance from the source were illustrated. Particular emphasis is placed on strong channeling. [Work supported by Naval Research Laboratory 6.1 program (element 601153N) with technical management provided by NRL-SSC.]

Session 5pSP

Speech Communication: Applications of Speech Perception Research in Disordered Populations

Brad S. Rakerd, Chair

Department of Audiology and Speech Science, Michigan State University, East Lansing, Michigan 48824

Contributed Papers

1:00

5pSP1. Speech perception without audition. Lynne E. Bernstein (Ctr. for Auditory and Speech Sci., Gallaudet Univ., 800 Florida Ave., N.E., Washington, DC 20002) and Marilyn E. Demorest (Univ. of Maryland, Baltimore County, Baltimore, MD 21228)

Most knowledge about speech perception is within the framework of studies concerned with acoustic-phonetic stimulus attributes. Vision is known to affect speech perception but is typically considered only as a supplement to hearing. The literature contains assertions that only 30% of words are recognized in lipreading/speechreading and that hearing subjects are more accurate than deaf subjects. Observations of expert visual speech perception among some profoundly deaf subjects [L. E. Bernstein *et al.*, *J. Acoust. Soc. Am.* **90**, 2971-2984 (1991)] promoted questioning of the conventional wisdom. A normative study of lipreading was therefore conducted involving 96 hearing and 72 deaf young adults. The two populations differed significantly across nonsense syllable, word, and sentence identification measures. The most accurate subjects were from within the deaf population. Results that lipreading in some deaf subjects is comparable to listening to speech in noise by hearing subjects. Expert lipreading demonstrates that speech perception does not require audition. Speech perception appears to be fundamentally the perception of linguistic entities not acoustic-phonetic attributes. Theories of speech perception for which generality is claimed need either to account also for visual speech perception or need to limit their claims to generality.

1:15

5pSP2. Stimulus materials for audio-visual studies of attention and speech perception by the hearing impaired. Brad Rakerd (Dept. of Audiol. and Speech Sci., Michigan State Univ., East Lansing, MI 48824) and Philip F. Seitz (Gallaudet Univ., Washington, DC 20002)

Individuals with sensorineural hearing impairment have been shown to devote a larger than normal share of their attention to speech processing when listening in situations that afford access to audio speech cues only. Attentional commitments may prove to be more nearly normal if both audio and visual speech cues are available. To test this possibility, a studio-quality videotape has been developed for use in primary-task and dual-task studies of attention and speech perception. Markers on the tape make it possible to maintain coordination with a computer. The timing of speech events is specified to within a few milliseconds, potentiating subject reaction time measurements. There are three sets of stimulus materials: (i) 30 to 45s samples of connected discourse; (ii) triplets of phonetically balanced monosyllabic words, arranged in an ABX format for discrimination experiments; and (iii) randomized lists of words and phonetically matched nonwords for lexical decision experiments. Copies of the videotape, along with supporting software and other test materials, are available upon request. [Work supported by NIH-NIDCD.]

1:30

5pSP3. Evaluating the articulation index for auditory-visual consonant recognition. Ken W. Grant and Brian E. Walden (Walter Reed Army Med. Ctr., Army Audiol. and Speech Ctr., Washington, DC 20307-5001)

The ANSI standard for calculating the articulation index [ANSI S3.5-1969 (R1986)] includes a procedure for estimating the effects of visual cues on speech intelligibility. This procedure assumes that listening conditions with the same auditory articulation index (AI_A) will have the same auditory-visual AI (AI_{AV}) regardless of the spectral composition of the signal. In contrast, other studies have suggested that the redundancy between A and V speech cues might be a better predictor of AV performance than either the AI_A or the overall (e.g., percent correct) auditory recognition score. In the present study, the ANSI procedure is evaluated by measuring A, V, and AV consonant recognition under a variety of different signal-to-noise and bandpass-filtered speech conditions. The results indicate that auditory conditions having the same AI_A do not necessarily result in the same AI_{AV} , and that low-frequency bands of speech tend to provide more benefit to speechreading than high-frequency bands of speech. Analyses of the auditory error patterns produced by the different filter conditions showed a strong negative correlation between the degree of A and V redundancy and the AV benefit obtained. These data indicate that the ANSI procedure is inadequate for predicting AV consonant recognition performance under conditions of severe spectral shaping. [Work supported by NIH Grant No. DC00792.]

5pSP4. Abstract withdrawn.

5pSP5. Relationship between identification errors and spectral details of naturally produced diphthongs. Anna K. Nabelek, Zbigniew Czyzewski, and Hilary Crowley (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, 457 South Stadium Hall, Knoxville, TN 37996-0740)

In an earlier study [A. K. Nabelek *et al.*, *J. Acoust. Soc. Am.* **92**, 1228–1246 (1992)], identification errors were collected for 15 English vowels, monophthongs, and diphthongs. The vowels were produced in a /b-t/ context by six male talkers who uttered five tokens of each vowel. There were 20 normal-hearing and 20 hearing-impaired subjects. Listening conditions were quiet, noise, and reverberation. Large differences in number of errors were observed among the 30 tokens of each vowel. The goal of the current study was to determine relationships between the spectral details of the diphthongs /aɪ, əɪ, əu/ and identification errors. Spectral analyses indicated that all three diphthongs had two distinct segments. In the first segment *F1* and *F2* were relatively steady state and in the second segment *F2* changed over time. For each token of each diphthong, frequencies and levels of *F1* and *F2* were traced along the duration of the diphthong. The results of this analysis seem to indicate a relationship between the number of identification errors for these diphthongs and the relative levels of the *F2* transition segment. Examples of highly identifiable and high confusable diphthongs will be shown. [This work was supported by NIDCD.]

2:00

5pSP6. Mutual dependence of octave-band contributions to speech intelligibility. Herman J. M. Steeneken (TNO Inst. for Human Factors, P.O. Box 23, 3769ZG, The Netherlands)

Current objective measures for predicting the intelligibility of speech assume that this can be modeled as a simple addition of the contributions from individual frequency bands. The articulation index (AI) and speech transmission index (STI) are based on this assumption. There is evidence that the underlying assumption of mutually independent contributions is not valid and may lead to erroneous prediction for conditions with a limited frequency transfer, with spectral gaps or with spectrally localized masking. An experiment was performed focused on the contributions from individual frequency bands. For this purpose the speech signal was subdivided into 7 octave bands with center frequencies ranging from 125 Hz to 8 kHz. For 26 different combinations of 3 or more octave bands the CVC-word score (consonant-vowel-consonant, nonsense words) was obtained at three signal-to-noise ratios. For an improved prediction of the observed intelligibility, a revised model was designed that accounts for mutual dependency between adjacent octave bands by the introduction of a redundancy correction (second order cross-product terms). A similar model was found for male and female speech. The data also provided a verification of the relation between signal-to-noise ratio and the contribution to the STI.

2:15

5pSP7. The separate contribution of head-shadow and binaural interactions to directional hearing in noise. Michael J. Nilsson, Sriram Jayaraman, and Sigfrid D. Soli (House Ear Inst., 2100 West Third St., Los Angeles, CA 90057)

A replication of Bronkhorst and Plomp [*J. Acoust. Soc. Am.* **83**, 1508–1516 (1988)] was undertaken to study the separate contributions of interaural level differences (ITDs); produced by head shadow, and interaural time differences (ITDs), which produce binaural interactions, to directional hearing in noise. Digital filtering of speech and noise signals was performed using head-related transfer functions (HRTFs) measured on a KEMAR mannequin to simulate sound field conditions under headphones. Speech materials from the hearing in noise test (HINT) were presented at a simulated 0-deg azimuth, while a noise signal matched to the spectral shape of the speech was filtered to include the ILDs and ITDs from either a 0-, 90-, or 270-deg presentation, including mixed combinations, e.g., 0-deg ILDs with 90-deg ITDs. Re-

sults from 19 normal hearing subjects showed the ILDs alone benefited the listener slightly more [3.97-dB improvement in sentence speech reception threshold (sSRT) versus a 0-deg condition] than ITDs alone (3.22-dB improvement in sSRT), as compared to 6.21-dB improvement in sSRT when both ILDs and ITDs are present. If the sSRT is calculated relative to the noise level in the shadowed ear, the ITDs alone produced the same sSRT as ITDs and ILDs together. Data from hearing impaired listeners will also be presented.

2:30–2:45 Break

2:45

5pSP8. Effects of hearing differences on encoding and comparison information-processing stages. Philip F. Seitz (Ctr. for Auditory and Speech Sci., Gallaudet Univ., Washington, DC 20002-3695), Brad Rakerd (Michigan State Univ.), and Paula E. Tucker (Gallaudet Univ.)

Reaction times to spoken digits presented in a speeded memory scanning procedure were measured for groups of listeners with normal hearing ($N=24$) and with congenital or early-onset sensorineural hearing losses ($N=12$). Separate groups of 12 normal-hearing listeners were tested under conditions of good stimulus quality (no distortion, high SNR) versus poor stimulus quality (low-pass filtered, low SNR). The speeded memory scanning procedure allows total reaction time to be decomposed into “encoding” and “comparison” components which correspond to separate stages in a model of human information processing. At issue here is whether long-term effects of hearing loss, such as possible deficits in phonological and/or lexical representation of spoken language, lead to unusual processing costs at either the encoding or comparison stage. Experimental results suggest that impaired listeners incur the same encoding costs as normal-hearing listeners presented with poor-quality stimuli. However, comparison costs for the impaired listeners are no higher than those for the normal-hearing listeners presented with good-quality stimuli, and are unexpectedly lower than those for the normal-hearing listeners presented with poor-quality stimuli. [Work supported by NIH-NIDCD.]

3:00

5pSP9. Speech recognition and frequency discrimination with the Ineraid and CIS cochlear prostheses. M. F. Dorman (Dept. of Speech and Hear. Sci., Arizona State Univ., Tempe, AZ 85287-0102)

Measures of frequency discrimination and speech understanding were made for cochlear implant patients using the Ineraid, analog signal processor and for patients using the continuous interleaved sampling (CIS) processor. For patients using the Ineraid, measures of frequency discrimination are significantly correlated with performance on tests of speech understanding. The correlation coefficients range from 0.60 to 0.82 depending on the frequency domain of the discrimination test and on the nature of the test of speech understanding. When Ineraid patients were fitted with a CIS processor, speech understanding improved significantly. An improvement in frequency discrimination may be the principal factor in the high word recognition scores achieved by patients using the CIS processor. [Research supported by NIDCD.]

3:15

5aSP10. Speaker-independent recognition of children's words with minimal phonemic contrast. Diane Kewley-Port, Bill Mills, and Jonathan Dalby (Commun. Disord. Technol., 205 S. Walnut, Bloomington, IN 47404)

Speaker-independent speech recognition can be used in speech training applications to provide feedback on phonemic contrasts. Our application is drill for normal-hearing misarticulating children with target-substitution errors (e.g., /r/→/w/ or /s/→/θ/). Six word pairs incorporating eight common misarticulations in word initial, medial, and final positions were repeated twice for a total of 192 words. These phonemic contrasts are extremely challenging for most recognizers. Sev-

enteen children in third to fifth grade repeated the word list once. Two other children and an adult repeated it seven times to provide speaker-dependent comparisons. The Scott SIR 20 recognizer was evaluated. Speaker-independent templates were made in round-robin fashion, leaving out one child each time for testing. The overall recognition rate for the 16 tests was 75.3% correct, with a range of 85.3% for the best (/s-f/) phonemic contrast, to 65.5% for the worst (/d-δ/). Human

recognition of a subset of the words was also shown to vary across phonemic contrasts. Analyses of the linguistic content of the word pairs and comparisons with the speaker-dependent recognition results indicate that speaker-independent recognition is accurate enough for some minimal phonemic contrasts to be used in speech drill. [Work supported by NIDCD.]

FRIDAY AFTERNOON, 8 OCTOBER 1993

COLUMBINE ROOM, 1:00 TO 4:00 P.M.

Session 5pUW

Underwater Acoustics: High-Frequency Acoustics in the Upper Ocean

Ralph R. Goodman, Chair

Applied Research Laboratory, Pennsylvania State University, State College, Pennsylvania 16804

Chair's Introduction—1:00

Invited Papers

1:05

5pUW1. Ocean surface processes affecting high-frequency acoustics. W. Kendall Melville (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0213)

Ocean surface processes affecting high-frequency acoustics are reviewed. The generation, propagation, and attenuation of high-frequency sound at and near the ocean surface is of interest for many current problems in acoustics and its applications in oceanography. While models of surface scattering by short surface gravity waves and even shorter gravity-capillary waves can be formulated, the scatterers are not homogeneously distributed on the surface. Nonlinear interactions with longer waves and swell, and direct forcing by the wind lead to modulations of the scatterers that are poorly understood. Nonlinear effects may not be weak, with the surface slope becoming infinite in breaking waves. Breaking waves also generate sound as the entrained air breaks up into smaller bubbles that radiate as they relax towards their equilibrium shape. The bubble plumes and clouds scatter and attenuate sound but field measurements of the acoustic properties of the bubble layer near the surface vary significantly. Bubble clouds may align with coherent Langmuir circulations. The surface layers may also be regions of strong temperature gradients (with concomitant sound-speed gradients), which are broken up by thermal convection and breaking waves. Natural and man-made surface slicks can suppress the waves and thereby modify both active and passive acoustics. These and other upper ocean processes will be reviewed.

1:30

5pUW2. High-frequency sea surface reverberation: An overview. Suzanne T. McDaniel (HC-01 Box 62, Spruce Creek, PA 16683)

High-frequency sea surface reverberation is reviewed from an historical perspective starting with the first comprehensive set of measurements made during World War II. Basic theories to predict reverberation due to scattering from resonant subsurface microbubbles and sea surface roughness are also reviewed as well as the environmental inputs to these theories: microbubble distributions and the ocean surface wave-number spectrum. Because resonant microbubbles absorb as well as scatter acoustic energy, a saturation effect is predicted, that is, when the bubble density increases beyond that needed to produce saturation, the backscatter remains constant. The reverberation level in this limit is a unique feature of scattering from resonant microbubbles permitting a clear identification of this scattering mechanism. Comparison of theoretical predictions with representative acoustic data demonstrates that reverberation at high grazing angles is due to rough surface scattering, while at lower grazing angles it is governed by scattering from resonant microbubbles. Large variations in backscattering strength are apparent for measurements performed at the same grazing angle, frequency, and wind speed, with reverberation levels in coastal waters being an order of magnitude higher than those in the open ocean. Although the physical mechanisms responsible for high-frequency reverberation are well understood, the relationship between backscattering variability and environmental factors is not.

1:55

5pUW3. High-frequency forward scattering from the sea surface: Theory and experiments. Peter H. Dahl (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

Intensity and coherence properties of high-frequency sound forward scattered from the sea surface are discussed. Propagation of high-frequency acoustic energy often involves a sea surface bounce path. In deep water, and for many sonar acquisition geometries, a single surface bounce may be the best path to the target, because the direct path has faded owing to the sound speed structure. In shallow water, interaction with both the sea surface and bottom is often the rule. For frequencies above 10 kHz, incoherent surface scattering in the forward direction will usually dominate over coherent reflection because of the large sea surface roughness. The mean total energy of the scattered field is reduced by the extinguishing effects of the near-surface bubble field through which surface bounce paths must traverse. Coherence properties of the scattered field, e.g., spatial and temporal, can

be related to the sea surface slope distribution and are also influenced by near-surface bubbles. Performance of detection and classification systems will depend on coherence properties, and in particular their degradation as a function of sea surface environmental conditions. Published theories on acoustic scattering from the sea surface in the high-frequency limit are discussed in the context of recent experimental measurements of sea surface forward scattering performed by the Applied Physics Laboratory in both inland water (Whidbey Island, Puget Sound) and open ocean conditions (R/P FLIP off California coast).

2:20

5pUW4. High-frequency bistatic ocean surface scattering. Richard Lee Culver (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16803) and Peter Neuman (Planning Systems, Inc., McLean, VA 22102)

Understanding and being able to predict high-frequency (10- to 100-kHz) acoustic scattering in the ocean for bistatic geometries (transmitter and receiver not collocated) has become increasingly important in recent years. For example, bistatic surface and bottom scattering must be understood in order to predict the angular dependence or spatial coherence of acoustic signals which have propagated through a shallow water channel. A high-frequency ocean surface scattering experiment conducted by Dahl and Jessup in 1992 using FLIP utilized an omnidirectional source and a line array receiver to measurement out-of-plane scattering at wind speeds up to 7 m/s. Estimates of low grazing angle bistatic scattering strength have been derived from these measurements, and are shown to compare favorably to theory developed by S. T. McDaniel. The McDaniel model utilizes composite-roughness theory to predict surface shape effects and includes attenuation and isotropic scatter by subsurface resonant bubbles. Bubble scattering is predicted to be the dominant mechanism at medium to high wind speeds and low grazing angles, and the FLIP experiment data offer evidence that this is the case.

Contributed Papers

2:45

5pUW5. High-frequency forward scattering from Gaussian spectrum, pressure release, corrugated surfaces: Catastrophe theory modeling. K. L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), J. S. Stroud, and P. L. Marston (Washington State Univ., Pullman, WA)

Exact integral equations for the acoustical pressure scattered from a rough pressure release surface can be written down and solved numerically. However, analytical head way into the problem, gained at the expense of various approximations, can lend physical insight not easily obtained numerically. This, in part, motivated the work to be discussed. In particular, a high-frequency approximation will be presented for forward scattering from Gaussian spectrum, pressure release, corrugated surfaces. The analysis is most directly applicable to forward scattering from an ocean surface dominated by swell. The presentation uses ideas and results from catastrophe theory [M. V. Berry, "Waves and Thom's Theorem," *Advan. Phys.* 25, 1-26 (1976); P. L. Marston, *Physical Acoustics* (Academic, New York, 1992), Vol. 21, pp. 1-234] to include diffraction. Catastrophe theory allows one to write down the scattered pressure in terms of a finite set of diffraction catastrophes and stationary phase contributions. The catastrophe theory results will be compared to numerical integration results for an individual rough surface to demonstrate their validity and the insight they supply. Both steady state and pulse comparisons will be made. Evidence will be presented that this "high-frequency" approximation is usable down to frequencies where the wavelength is twice the rms roughness of the surface. [Work supported by Office of Naval Research.]

3:00

5pUW6. High-frequency forward scattering from Gaussian spectrum, pressure release, corrugated surfaces: Experiment and comparison with catastrophe theory. J. S. Stroud, P. L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814), and K. L. Williams (Univ. of Washington, Seattle, WA)

A single realization of a Gaussian spectrum surface, rms roughness 1.5 cm, correlation length 10 cm was manufactured out of styrofoam. This surface provided a pressure release, corrugated surface for an underwater, forward scattering investigation. Omni-directional source and receiver were used in the frequency range of 100 to 300 kHz. Short pulses were used to allow isolation of individual contributions to the scattered field. These individual contributions were then classified using catastrophe theory [M. V. Berry, "Waves and Thom's Theorem," *Advan. Phys.* 25, 1-26 (1976); P. L. Marston, *Physical Acoustics* (Academic, New York, 1992), Vol. 21, pp. 1-234]. Scans at a constant

distance from the mean of the rough surface were used to determine the peak forward scattered pressure. Composites of these types of scans are shown and discussed using catastrophe theory. The catastrophe theory simulation is shown to be a valid approximation to the wavefield for a wide range of frequencies. [Work supported by Office of Naval Research.]

3:15

5pUW7. High-frequency propagation in a turbulent shallow water environment. Daniela DiIorio and David M. Farmer (Inst. of Ocean Sci., P.O. Box 6000, Sidney, BC V8L 4B2, Canada)

A high-frequency (67-kHz) acoustic propagation experiment conducted in a shallow tidal channel provides an opportunity for comparison of observed acoustical scintillations with available theory. The tidal current ensures that the acoustic path runs through a turbulent boundary layer at all times. Analysis of log-amplitude, phase, and phase-difference signals shows close agreement with the weak scattering theory of Tatarski using the Kolmogorov turbulence model together with the Taylor hypothesis of "frozen" advected turbulence. By combining acoustic data with direct measurement of temperature and salinity fluctuations using *in situ* sensors, the contribution of turbulent velocity fluctuations to the scintillation signal is evaluated. The results show that turbulent velocity fluctuations rather than temperature and salinity variability are the dominant source of observed acoustic fluctuations, leading to estimates of turbulent energy dissipation in the range $\epsilon \approx 10^{-5} \text{ m}^2 \text{ s}^{-3}$.

3:30

5pUW8. Sound propagation through a bubble screen. Christopher M. Hobbs and Ali R. Kolaini (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677)

Results of a laboratory experiment to characterize the sound propagation through a bubble screen are described. A bubble screen of various void fractions was positioned in the middle of an anechoic water tank. The cloud was driven using a broadband source and the acoustic pressure was measured on both sides of the layer. The results have revealed that there is significant attenuation near the resonance frequency of the bubbles contained within the screen. When the source frequency is near that of one of the "collective-oscillation" frequencies of the screen, sound pressure amplification may occur. In this paper the new findings corresponding to the collective mode amplification, high-frequency attenuation, as a function of the screen's physical properties are discussed. [Work supported by ONR.]

5pUW9. Low-frequency acoustic emissions by impacting transient cylindrical water jets in fresh and salt water. Ali R. Kolaini (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, MS 38677), Ronald A. Roy (Univ. of Washington, Seattle, WA 98105), and David L. Gardner (NOAA, PMEL, Seattle, WA 98115)

The impact of a jet of water onto a still water surface results in the entrainment of large amounts of air and the eventual formation of a bubble plume. The densely populated bubble plumes were generated by dropping a fixed volume of water, held in a cylindrical container, onto a still-water surface. The detached bubble plume, which is roughly spherical in shape, undergoes volume pulsations and radiates relatively

large-amplitude, low-frequency sound. The results of laboratory study of the noise produced by this process were reported previously by Kolaini *et al.* [*J. Acoust. Soc. Am.* **89**, 2452–2455 (1991)]. In this presentation, a field study of noise produced by this process in both fresh water (Lake Washington) and salt water (the Puget Sound) will be described. Studies of acoustic emissions from transient bubble plumes as a function of cylinder parameters will be described, with specific attention devoted to a comparison of results obtained in salt and fresh water. The measurements indicate that there is a correlation between the acoustic intensity radiated from bubble plumes and the total potential energy of the water jet. [Work supported by ONR.]

FRIDAY AFTERNOON, 8 OCTOBER 1993 (TO BE ANNOUNCED) ROOM, 1:00 TO 2:30 P.M.

Session 5pAA

Architectural Acoustics: Vern O. Knudsen Distinguished Lecture

David Lubman, Chair

David Lubman & Associates, Westminister, California 92683

Chair's Introduction—1:00

Invited Paper

1:05

5pAA1. Quantifying musical acoustics through audibility. David H. Griesinger (Lexicon, 100 Beaver St., Waltham, MA 02154)

This paper proposes a set of measures of musical acoustics derived from a study of binaural hearing and speech perception. The measures are tested on analyzing data from unoccupied and occupied halls, data from several halls which include electronically variable acoustics capable of a wide range of adjustment, and data from a binaural synthesizer of hall acoustics. The new measures include the 500-ms reverberant level (L500), early (0–60 ms) reverb time (ERT), middle (0.1–0.5 s) reverb time (MRT), late (0.5+) reverb time (LRT), early binaural fraction (EBF), and middle binaural fraction (MBF). L500, a measure of reverberant level, is related to early decay time (EDT). Optimum L500 is found to be high in the range of musical fundamentals, and low in the region of maximum speech information. ERT is in part a measure of the localizability of the sound, which is related to its clarity. MRT is particularly significant whenever the decay curve is not a simple exponential. It is found that when MRT is much longer than ERT the sound can be both clear and reverberant at the same time. The binaural measures are found to be similar to IACC, but have advantages at frequencies below 300 Hz. Some implications of this research on optimum hall shape and surface treatment are discussed.

2:00

DISCUSSION