

Low-frequency pressure wave propagation in liquid-filled, flexible tubes. (A)

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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 MORNING, 11 MAY 1992

SALON G, 8:00 A.M. TO 12:15 P.M.

Session 0AO

Acoustical Oceanography: Acoustical Studies of Upper Ocean Dynamic Processes I

Andrea Prosperetti, Chair

Department of Mechanical Engineering, Johns Hopkins University, Baltimore, Maryland 21218

Chair's Introduction—8:00

Invited Papers

8:05

0AO1. Low-frequency sound radiation and scattering from bubble clouds. William M. Carey (Defense Appl. Res. Project Agency, Arlington, VA 22202)

Recent experimental evidence has shown that when wave breaking occurs, low-frequency (LF~200 Hz) sound is produced and LF scatter has a different characteristic than expected from rough sea surface scattering. These effects have been attributed to the bubbles produced during wave breaking, which are convected to depth by the breaking turbulence, vorticity and Langmuir circulation as observed by Thorpe [S. Thorpe, *Oceanic White Caps*, edited by E. Monahan and G. MacNiocaill (Reidel, Boston, 1986), pp. 57–58]. While the radiation and scattering characteristics at frequencies greater than 1 kHz are explained by incoherent scatter from the observed bubble size and space distributions the lower frequency phenomena are not easily explained. However if bubble plumes and clouds produced in the wave breaking have appreciable volume fractions ($> 10^{-5}$), then LF sound radiation and scattering can be explained by classical theories. This paper reviews the scattering and radiation from bubble clouds in water as a function of volume fraction. When the cloud is compact, coherent and collective scatter are shown to occur. The natural frequency of radiation is shown to be described by a modified Minnaert result while the backscatter target strength is described by the first-order volume mode. These analytical results agree with experimental sound radiation and scatter measurements. Finally, the collective radiation of bubble plumes and clouds is discussed as a possible explanation of the observed ocean low-frequency scattering and radiation phenomena. [Work sponsored by ONR 11250A and NUSC IR.]

0A02. Low-frequency backscatter from dense submerged bubble clouds. Ronald A. Roy (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), William Carey (Defense Appl. Res. Project Agency, Arlington, VA 22202), Michael Nicholas, Lawrence A. Crum, and Jeff Schindall (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677)

The frequency-dependent characteristics of the acoustic backscatter from submerged bubble clouds was studied using both conventional and parametric sound sources. The clouds, all of which possessed a characteristic length scale of 0.5 m, a void fraction of order 10^{-3} , and a mean bubble radius of approximately 1.5 mm, were generated 39.2 m beneath the surface of a freshwater lake (Lake Seneca, NY) using an array of hypodermic needles driven with pulsed flow of compressed air. Backscattering target strengths were measured for frequencies ranging from 300 Hz to 14 kHz. Results obtained using conventional and parametric sources were comparable, with target strengths falling in the +1 to -10-dB range. Several peaks were observed in the target strength versus frequency data. Low-frequency (less than 1 kHz) results exhibit elevated target strengths throughout, with evidence of a broad peak at approximately 500 Hz. Results obtained over the entire range of frequencies will be compared with theoretical predictions based on coherent, resonance scattering (> 1 kHz) and coherent, off-resonance scattering (< 1 kHz) and collective bubble oscillations (< 1 kHz). [Work supported by ONR, ONT, AEAS, and NUSC/IR.]

0A03. Laboratory experiments on bubble-cloud oscillations. Lawrence A. Crum, Michael Nicholas, Ali Kolaini (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, Oxford, MS 38677), and Ronald A. Roy (Univ. of Washington, Seattle, WA 98105)

A variety of experiments that have been undertaken to investigate the acoustic characteristics of bubble clouds will be described. These experiments have involved both passive acoustic emission, in which the radiated noise is measured from a bubble cloud that is being produced, and active acoustic scattering, in which sound waves are scattered from a similarly produced bubble cloud after it has ceased to radiate emissions. It appears that the dominant acoustic characteristic of these clouds is that they readily emit, and can easily be stimulated into emitting, low-frequency sound by the collective oscillations of the bubbles within the cloud. This mode of oscillation [see papers by Prosperetti and by Carey and Browning in *Sea Surface Sound*, edited by B. R. Kerman (Kluwer Academic, Dordrecht, 1988)] appears to provide a physical mechanism for a broad range of ambient noise emissions in the ocean and may play an important role in low-frequency active scattering from the ocean surface. [Work supported by ONR and ONT.]

0A04. Two-dimensional imaging of bubble cloud distribution. David M. Farmer and Mark Trevorrow (Inst. of Ocean Sci., P.O. Box 1600, Sidney, BC V8L 4B2, Canada)

Observation of bubble cloud organization with sidescan sonars having fixed orientation is now a well established technique that has revealed aspects of bubble cloud behavior relevant to the study of low-frequency acoustic scattering, Langmuir circulation, and related topics. A difficulty with this approach is that the signal from a fixed sonar is limited to whatever drifts through the field of view, so that temporal and spatial evolution of the patterns must be inferred from the time-dependent, one-dimensional signal. Observations have now been obtained with an azimuthally scanning sonar that can acquire bubble cloud imagery every 30-40 s through 360° over a disk of radius approximately 300 m. Bubble cloud evolution is slow relative to this scanning rate, so that successive images form a "movie" of the evolving bubble cloud patterns. Such images can be used to determine their length and stability, and the way in which adjacent clouds amalgamate. [Work supported by ONR.]

0A05. Using acoustic scattering (high frequency) to predict acoustic scattering (low frequency). Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 40th St. NE, Seattle, WA 98105), David Farmer (Inst. of Ocean Sciences, Sidney, BC, Canada), and Svein Vagle (Bergen Sci. Ctr., Bergen, Norway)

Models of low-frequency reverberation from tenuous bubble clouds have made assumptions about the geometry of the clouds. Some of those assumptions are replaced with properties remotely measured using high-frequency scattering. This approach is applied to the measurements made during the CST-7 experiment, which is the first experiment including both high- and low-frequency measurements. Also discussed is the use of ocean information obtained from the high-frequency scattering in addition to the geometry of the clouds to help infer low-frequency reverberation characteristics. [Work supported by the Office of Naval Research.]

Break

10:25

0A06. Fractal characterization of the near-surface oceanic bubble layer from deep-ocean reverberation measurements. Kenneth E. Gilbert (Graduate Program in Acoustics and Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804) and Lintao Wang (Natl. Ctr. for Phys. Acoust., University, MS 38677)

At high sea states and low grazing angles, it has been hypothesized that surface reverberation is dominated by backscatter from the bubble layer beneath the sea surface. Assuming the hypothesis is true, a simple stochastic bubble-layer backscatter model has been used to analyze surface reverberation data from four different deep-ocean areas. In every area, the inferred wave-number spectrum for the horizontal structure of the bubble layer is an inverse power law of the form $P(K) = AK^{-\beta}$, where A is a constant, K is the horizontal wave number of the structure, and $3 \lesssim \beta \lesssim 4$. The existence of power-law wave-number spectra indicates that the horizontal structure of the bubble layer is fractal. Using $D = 4 - \beta/2$ for the fractal dimension, $2 \lesssim D \lesssim 2.5$ is obtained for measurements analyzed so far. Fractal structure in the bubble layer implies that horizontal inhomogeneities exist on a wide range of scales and possess scale invariance. Hence, on the average, the medium looks the same at all scales between some "inner scale" and "outer scale." (Presently, the inner and outer scales are not known.) In this paper some physical processes are considered that could generate fractal structure in the bubble layer and some experimental means are suggested for identifying the processes. [Work supported by ONR.]

10:50

0A07. A critique on the relationship between backscattering and environmental measurements. R. R. Goodman (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

In the paper by Gilbert that was presented earlier in this session it was shown that, by the use of the first-order Born approximation and a plausible assumption about the vertical distribution of bubbles, backscattering strength measurements in several experiments yield horizontal wave-number power spectra for bubble densities of the form $P(K_h) = AK_h^{-\beta}$. The values of β were found to lie approximately between 3 and 4. Since a given value of β requires that the oceanographic processes produce a bubble distribution that obeys, on the average, a specific scaling invariance, it is interesting, then, to see if there are any correlations between the oceanographic/atmospheric observations and the values of β for the experiments that were analyzed. It is the purpose of this paper to examine the results of each of the experiments from the standpoint of acoustics and environmental observations in order to seek "clues" to the correlation of β with oceanographic/atmospheric observations. It will be clear that none of the experiments contained measurements that are sufficient to yield specific connections between the environmental measurements and the acoustics.

Contributed Paper

11:15

0A08. Enhanced backscattering from bubble cloud distributions on the ocean surface. Andrea Prosperetti and Kausik Sarkar (Dept. of Mech. Eng., The Johns Hopkins Univ., Baltimore, MD 21218)

It has been shown in earlier studies [Prosperetti, Lu, and Kim; Sarkar and Prosperetti, both submitted to *J. Acoust. Soc. Am.*] that bubble clouds produced by breaking waves at the ocean's surface can explain the unexpectedly high backscattering levels observed experimentally by Chapman and Harris [*J. Acoust. Soc. Am.* **34**, 1592

(1962)] and others at low grazing angles. Gas volume fractions of the order of 1%, linear dimensions of the order of 1 m, and surface coverage of the order of 1% (the latter of which agrees with the experimentally measured values for 10 m/s winds) are sufficient to give an excellent match of the data as a function of frequency in the range 0.1–2 kHz and wind speeds from 5 to 25 m/s. In the previous work the clouds were treated as independent scatterers. In the present work the previous results are refined to include lowest-order multiple scattering effects along the lines of Foldy [*Phys. Rev.* **67**, 107 (1945)] and Biot [*J. Acoust. Soc. Am.* **44**, 1616 (1968)]. [Work supported by ONR.]

Panel Discussion

PANEL MODERATOR: Andrea Prosperetti

**PANEL MEMBERS: William M. Carey
Lawrence A. Crum
David M. Farmer
Kenneth E. Gilbert
Ralph R. Goodman
Frank S. Henyey
Ronald A. Roy**

Session 1AO

Acoustical Oceanography: Acoustical Studies of Upper Ocean Dynamic Processes II

Ralph R. Goodman, Chair

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

Chair's Introduction—1:30

Invited Papers

1:35

1AO1. The proposed Knight Inlet surface backscattering experiment. T. Ewart, D. Farmer, F. Henyey, D. Jackson, P. Kaczowski, L. Olson, and E. Thorsos (Univ. of Washington, Seattle, WA 98105)

An experiment is planned for the 93–95 time frame to measure surface backscattering at 400 Hz using a “pencil beam” source and receiver currently under development. The acoustics experiment will be conducted simultaneously with surface zone environmental measurements (notably, surface height statistics and bubble densities) contributed by various scientists in the air–sea acoustics program. Initial operation in the fjord allows the equipment to be installed at low wind speeds, and the apparatus can be connected by cable to the protected waters of a small bay. In this way operation of the equipment over a wide variety of wind conditions can be achieved safely prior to open ocean deployment. (High winds are common in the confined channel of the fjord.) As the principal goal, the apparatus will be used to determine the regime where rough surface scattering theory does not adequately describe the observations, and to determine the dominant mechanisms. In this scattering regime the measurements will focus on the relative importance of the various surface zone phenomena (bubbles in their different incarnations) predicted to yield an enhanced surface backscattering signature. The rationale and design of the experiment will be discussed. The experiment will provide the experience required before open ocean measurements can be undertaken.

2:00

1AO2. Wave breaking, turbulence, and related similarity laws—A review of wind-wave research at Tohoku University. Y. Toba, H. Kawamura, and N. Ebuchi (Dept. of Geophys., Tohoku Univ., Sendai, 980 Japan)

Research on wind waves made by our group at Tohoku University is reviewed with emphasis on wave breaking, turbulence, and the underlying similarity laws. Detailed dynamical processes related to wind waves, including ordered motions and turbulence in the air and water, have been investigated in a tank, using quantitative flow visualization techniques combined with the direct measurement of turbulent flows. Conspicuous events, such as airflow separation, formation of a high-vorticity layer at the wave crest, wave breaking, and bursting in air and water are found in the vicinity of the water surface. They are related to one another to explain momentum transfer from the airflow and turbulence generation, which may be associated with sound generation at the water surface. Similarity laws exist, which correspond to the relationship between waves and turbulence below and above the interface, describing the gross nature of the wind-wave field. A parameter $u_*^2/\nu\sigma_p$ is introduced to describe overall processes of wind-wave breaking, whitewater, and drop production. It is suggested that this parameter may describe also the overall intensity of bubble formation in the sea.

2:25

1AO3. The effect of the rough air–sea interface on scattering from tenuous bubble clouds. Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Henyey has developed a theory for low-frequency scattering from tenuous bubble clouds [F. S. Henyey, *J. Acoust. Soc. Am.* **90**, 399–405 (1990)]. The bubbly water is treated as an effective medium, described with a spatially varying index of refraction, and the scattering is found using a perturbation theory approach. This model applies to bubble clouds with sufficiently low air volume fraction that buoyancy forces are negligible; the bubble clouds are thus referred to as tenuous. Henyey simplified his analysis by assuming that the air–sea interface is flat. Here the effect of including surface roughness is examined for the 2-D problem (1-D surface roughness), but otherwise Henyey's approach is followed. The integral equation method is used to obtain the exact acoustic fields near the rough surface, which are then employed in the

bubble scattering calculations using perturbation theory. Results comparing bubble scattering with rough and flat sea surfaces show that the average scattered intensity is not much affected by surface roughness, but the statistical distribution of intensities can be significantly affected, leading in some cases to a hightailed intensity distribution (spikes) when the surface is rough. Implications for the time dependence of the scattering and for the Doppler spectrum will also be discussed. [Work supported by ONR.]

2:50

1AO4. Acoustic daylight: Imaging the ocean with ambient noise. Michael J. Buckingham, Broderick V. Berkhout (Marine Phys. Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0213), and Stewart A. L. Glegg (Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

For many years the principal means of probing the ocean using sound has been through the use of either "active" or "passive" techniques. With an active system, an object is illuminated by a pulse of sound and its presence inferred from the echo it produces, whereas the passive approach involves simply listening for the sound that the object itself emits. Here a new method of employing sound in the ocean is reported, which is neither passive nor active. It relies on the naturally occurring, incoherent ambient noise field in the ocean as the sole source of acoustic illumination. By analogy with daylight in the atmosphere, ambient noise scattered from an object in the ocean can be focused onto an image plane, and from this acoustic image, with appropriate signal processing, a visual image can be produced on a television-type monitor of the object space in the ocean. To test this concept, several simple experiments have been conducted in the ocean off Scripps pier, with a single parabolic reflector acting as an acoustic lens, which confirm that objects illuminated only by ambient noise can indeed be "seen" over a frequency band between 5 and 50 kHz. [Work supported by ONR.]

3:15-3:30

Break

3:30

1AO5. Bubble plume echoes from convergence zone ranges. B. E. McDonald, G. J. Orris, and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Surface reverberation from bubble plumes at long ranges (one or two convergence zones) is modeled by combining a Born approximation scatter model [McDonald, *J. Acoust. Soc. Am.* **89**, 617-622 (1990)], which has given agreement with surface backscatter data at close ranges (much less than a convergence zone), and an efficient method for coupling the scattering and propagation aspects of the problem [Collins and Werby, *J. Acoust. Soc. Am.* **85**, 1895-1902 (1989)]. A signal from a point source near the surface is propagated through a realistically refractive ocean to convergence, where it incoherently insonifies a set of bubble plumes. These are represented as vertically aligned cylindrical patches containing a sound-speed defect that diminishes exponentially with depth. For low frequencies (on the order of 100 Hz), the hypothesis that backscatter is due almost entirely to deeply entrained bubbles, likely resulting from Langmuir circulation cells, is examined.

3:55

1AO6. Acoustic radiation due to surface wave breaking. Robert M. Kennedy (Naval Underwater Systems Ctr., AUTEC, West Palm Beach, FL 33402-7517) and Stewart A. L. Glegg (Florida Atlantic Univ., Boca Raton, FL 33431)

While wave breaking is continually occurring at the sea surface, its transient and sporadic nature makes it difficult to measure. Experimental results are presented that show how acoustic methods can be used as a remote sensor of this fundamental process. Sea surface-generated acoustic radiation (40 to 4000 Hz) is directly related to a quantitative measure of the boundary dynamics; i.e., the Toba variable. The frequency spectrum of the radiation remains remarkably unchanged over a wide range of environmental conditions but the correlation between the sound pressure level and the Toba variable undergoes an abrupt change when spilling breakers start to occur. Results support the use of acoustics to remotely measure the rate of energy being dissipated by wave breaking and the wavelength of the dominant gravity wave component. Theoretical studies have related the field measurements to analytical and laboratory results cited in the literature indicating that remote monitoring of the rate of occurrence and size distribution of "infant" (freshly entrained) bubbles may be possible if splashes on the surface do not radiate significant sound. Signal processing algorithms for the remote measurements discussed above are enhanced by eigenstructure analysis of the measured cross-spectral density matrix. [Work sponsored by ONR and NUSC.]

1A07. High-frequency surface scattering phenomena in high sea states. Kerry W. Commander (Coastal Systems Station, Panama City, FL 32407-5000)

The determination of acoustic scattering from a rough sea surface has received considerable attention in recent years. Knowledge of whitecap characteristics [J. Wu, *IEEE J. Oceanic Eng.* 17, 150–158 (1992)] has led to a better understanding of surface scattering phenomena [D. F. McCammon and S. T. McDaniel, *IEEE J. Oceanic Eng.* 15, 95–100 (1990)]. For this study a substantial amount of backscattering data was acquired with autonomous underwater vehicles running between 50 and 100 ft of the surface. These vehicles used a short pulse length sonar operating at 215 kHz. The use of a high-frequency sonar allows deep penetration of densely populated bubble plumes. Because of the high repetition rate, good spatial variation of backscattering strength from breaking waves has been obtained. Previous measurements by other scientists of bubble distribution functions in breaking waves allows an approximate conversion of scattering strength at the sonar frequency to most frequencies of interest. The database here consists of measurements made in deep water environments with seas ranging from sea state 0 to 3. The spatial variation of scattering strength due to the sea surface and bubble plumes from breaking waves is discussed as a function of wind speed.

Contributed Papers

4:45

1A08. Short pulse acoustic transmission through marine microbubbles. Alastair Cowley (Defence Res. Agency, Maritime Div., Portland, UK), Nicholas G. Pace, and Tony Campbell (Bath University, Avon, UK)

When a bubble is excited at or near its natural frequency it is a very effective scatterer and absorber of sound. In fact, at resonance, scattering and absorption cross sections are of the order of 1000 times greater than the geometric cross section for typical marine microbubbles. As with any resonant system, a bubble takes a finite albeit short time to ring up to steady state oscillation and continues to oscillate for a finite time after the driving force ceases. A number of studies have been carried out to attempt to describe how acoustic absorption and backscatter depend upon the duration of the driving force. Most notable of these are the backscatter measurement for short pulses in near surface seawater [Akulichev *et al.*, *Sov. Phys. Acoust.* 23, 177–180 (1986)] and the forward transmission laboratory measurements for short pulse lengths at 120 kHz [H. R. Suiter, *J. Acoust. Soc. Am.* 91, 1383–1387 (1992)]. Akulichev observed a decrease in acoustic backscatter for short pulses at three discrete frequencies, whereas Suiter found no corresponding enhancement in forward transmission at 120 kHz. This paper details a comprehensive study that measured acoustic attenuation through a well defined bubble cloud over the frequency range 20–200 kHz and for pulse lengths from 20 cycles down to a single cycle using a parametric transmitter. The experiment simulated the conditions for which a decrease in

attenuation with decreasing pulse length might have been expected. No effect was observed for two different but well defined bubble distributions. [Work supported by DRA Maritime.]

5:00

1A09. Nonlinear wind-wave effects in backscatter. R. H. Mellen and I. A. Leykin (Marine Sci. Inst., Univ. of Connecticut, Groton, CT 06340)

In backscatter experiments, nonlinear wind-wave effects have been observed in Doppler spectra and azimuthal asymmetry of scattering strength. Both effects have been ascribed to spatial skewness associated with nonlinear steepening of the wavefronts. Bicoherence analysis of wavegauge measurements in a laboratory flume supports this mechanism. However, standard statistical methods reveal only average properties. In a recent paper [N. E. Huang, *Nonlinear Water Waves Workshop*, Bristol, U. K. (1991)], Huang reported on the use of phase/time analysis of the analytic wave function to obtain additional information on local properties and analyzed wave-buoy data by this method. His results showed evidence of “wave groups” in addition to possible fractal properties, the spectra of the phase fluctuations being consistent with a self-similar process of fractal dimension ≈ 1.4 . Phase/time analysis of the laboratory flume data is reported here with similar results and this appears to support fractal behavior of the “wave groups.” Some of the implications for backscatter are considered.

5:15–6:00

Panel Discussion

PANEL MODERATOR: Ralph R. Goodman

PANEL MEMBERS: Michael J. Buckingham
Kerry W. Commander
Terry E. Ewart
Robert M. Kennedy
B. E. McDonald
Eric I. Thorsos
Y. Toba

Session 1ID

Tutorial on Digital Audio

Mauro Pierucci, Chair

*Department of Aerospace and Engineering Mechanics, San Diego State University,
San Diego, California 92182*

Chair's Introduction—7:00

Invited Paper

7:05

1ID1. Digital audio. Thomas G. Stockham, Jr. (Elec. Eng. Dept., Univ. of Utah, Salt Lake City, UT 84112)

This tutorial addresses an audience with no background in digital audio and builds a basic and fundamental understanding of the subject. The presentation covers a set of topics that define a digital audio chain from input to output. The topics are signal preparation, sampling, digitization, recording/correction, signal processing, editing, playback, error detection/correction, decoding, and signal reconstruction. This set of concepts provides substantial insight into digital audio across a broad spectrum of applications from the lowest quality voice to the highest quality music.

Session 2AO

Acoustical Oceanography: Acoustical Studies of Upper Ocean Dynamic Processes III

Terry E. Ewart, Chair

Applied Physics Laboratory, University of Washington, 1013 East 40th Street, Seattle, Washington 98195

Chair's Introduction—8:00

Invited Papers

8:05

2AO1. Upper ocean diagnostics during rainfall. Herman Medwin (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

Laboratory measurements have been made of the sound radiated by the complete size range of acoustically significant, single raindrops, falling at terminal speeds onto the surface of an anechoic water environment. This has provided the source information for the detailed analysis of many effects of light to heavy rainfall at sea. The three acoustically distinctive subranges are defined as "small" drops (equivalent diameter, 0.8–1.1 mm), "midsize" drops (1.1–2.2 mm), and "large drops" (> 2.2 mm). These subranges correspond to drops that are oblate ellipsoids, slightly flat-bottomed ellipsoids, and completely flat or concave-bottomed ellipsoids just before they hit the water surface. Their penetration through the air–water interface results in underwater spectra which, respectively, have peaks around 15 kHz, are very broadband, and have peaks that decrease from 10 to 2 kHz as the drop diameter increases from 2.2 to 5 mm. This knowledge of the acoustical sources at the surface has now made it possible to determine, during rainfall, such ocean parameters as: gas transfer by active microbubbles, bubble populations under the sea surface, rms ocean slope, raindrop size distribution, total rainfall rate, raindrop temperature, and even the type cloud and its base height from which the rain came. Some results will be presented. [Work supported by the ONR.]

8:30

2AO2. An inversion technique to measure rain using underwater sound. Jeffrey A. Nystuen and Leo H. Ostwald, Jr.¹ (Dept. of Oceanogr., Naval Postgraduate School, Monterey, CA 93943)

Laboratory measurements of the sound generated by individual raindrops striking at terminal velocity show distinct spectral signatures for drop sizes from 2.2–4.6 mm diameter. These spectral signatures form a basis of a linear inversion algorithm that allows the large drop size distribution in rain to be estimated from the measured underwater sound spectrum. As large drops comprise most of the water volume, the estimate of rainfall rate should be very good. Examples of both the forward problem—estimating the underwater sound spectrum given the drop size distribution—and the inverse problem—estimating the drop size distribution given the sound spectrum—will be presented and compared to field observations. [Work supported by NOARL and NPS.] ¹Lt., USN.

8:55

2AO3. Rain noise. John E. Ffowcs-Williams and Hugh C. Pumphrey (Eng. Dept., Cambridge Univ., Trumpington St., Cambridge CB2 1PZ, England)

This talk will review the progress made over the last 40 years in the understanding of the sound made by rain falling onto a water surface. Measurements were made in the early 1950's, but the story really begins when Franz [J. Acoust. Soc. Am. 31, 1080–1096 (1959)] showed that a raindrop can make sound in two different ways: by a water hammer at the moment of impact, and by entraining a bubble some tens of milliseconds after. Subsequent work has largely consisted of a long controversy over which of these effects was most significant at various frequencies. Franz was not really in a position to draw a valid conclusion as he had no good data for real rain, a situation which persisted until the mid-1980's, when Nystuen [J. Acoust. Soc. Am. 79, 972–987 (1986)] and Scrimger [Nature 318, 647–649 (1985)] separately discovered that the power spectrum of rain noise has a large peak at 14 kHz. Nystuen tried initially to explain the peak in terms of the initial impact sound, but one of the present authors [Pumphrey *et al.*, J. Acoust. Soc. Am. 85, 1518–1526 (1986)] showed that it is due to bubbles entrained by a process that was not known about before that date. Since then, this process has been extensively studied and is fairly well understood from an

experimental point of view, but the initial impact sound, although it has been modeled theoretically, is still a matter of debate for experimentalists. The present state of knowledge will be presented, and suggestions made for future work.

9:20-9:35

Break

9:35

2AO4. Distortion of bubble clouds by orbital motions in surface waves. Michael S. Longuet-Higgins (Inst. for Nonlinear Sci., UCSD, La Jolla, CA 92093-0402)

The acoustical properties of a bubble plume, both as an emitter and as a backscatterer of underwater sound, will be strongly influenced by its geometrical shape and dimensions, and by its distance from the sea surface. In this paper the distortion of a bubble plume due to both the periodic motion in a steep, irrotational wave and the Lagrangian-mean shearing motion associated with the Stokes drift are calculated. It is shown that the bubble plume is rotated and also brought closer to the free surface on the rear face of the wave. The cloud is furthest from the surface when it is immediately below the wave crest. That is neglecting the natural buoyancy of the plume. The dynamical effects of buoyancy and mixing can also be taken into account. [Work supported by ONR.]

10:00

2AO5. The dynamics and acoustics of breaking waves. W. K. Melville (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093 and R. M. Parsons Lab., MIT, Cambridge, MA 02139), E. Lamarre, and M. R. Loewen (R. M. Parsons Lab., 48-108, MIT, Cambridge, MA 02139)

In this presentation, recent progress is reviewed in the understanding of the relationship between the dynamics of breaking and the sound generated by breaking waves. Gently breaking waves entrain a small number of bubbles that may break up, resonate, and relax toward their equilibrium shape. Medwin and Daniel's (1990) measurements of the sound spectrum under gently spilling waves are shown to be consistent with a simple model of the acoustics based on the resonant relaxation of individual bubbles. More energetic breaking may lead to significant air entrainment with recent laboratory and field measurements showing void fractions of $O(10^{-1})$ or more. At these large void fractions the entrained air becomes dynamically significant and up to 50% of the energy lost from the surface wave field by breaking may be expected by entraining air against buoyancy forces. Bubble clouds of large void fraction may oscillate collectively and radiate sound at frequencies much lower than the resonant frequencies of the individual bubbles. Elsewhere at this meeting, measurements of low-frequency sound generated by 2-D and 3-D breaking waves are presented, which are consistent with collective oscillation frequencies predicted from independent measurements of the void fraction. Methods are reviewed by which acoustic measurements of breaking may prove useful in indirect measurements of the dynamics of breaking and its role in processes of air-sea interaction. [Work supported by ONR (Ocean Acoustics) and NSF (Physical Oceanography).]

10:25

2AO6. Nearsurface bubble density distribution due to wave breaking. Ming-Yang Su (Naval Res. Lab., Stennis Space Ctr., MS 39529-5004)

Several series of experiments have been conducted in the past 2 years on breaking of surface gravity waves and its subsequent air entrainment, which forms plumes of bubbles. These experiments were conducted using both electromagnetic and acoustic principles. The laboratory experiments were performed in a very large tank, in both fresh and salt water, and were used to quantify the significant effects of salinity on the production and evolution of bubbles due to wave breaking. From the open sea experiments, extensive measurements of the near-surface void fraction under moderate to high sea states yield some remarkable new studies of occurrence of breaking waves as well as the significant effect of wave age on such statistics. From both the laboratory and open ocean measurements, a universal size distribution of bubbles with radii ranging from 34 to 1200 μ due to surface gravity wave breaking has been observed.

10:50

2AO7. Temporal evolution of the bubble plumes generated by breaking waves: Acoustical implications. Edward C. Monahan and Qin Wang (Marine Sci. Inst., Univ. of Connecticut, 1084 Shennecossett Rd., Groton, CT 06340-6097)

Via a series of "tipping bucket" experiments in UConn's Whitecap Simulation Tank IV, the temporal evolution of the bubble plumes that result from spilling breaking waves in fresh water and sea water has been investigated. The saltwater bubble plume in its early stages is characterized by many more bubbles than are encountered in a freshwater plume in the same stages. The peak in the early salt-water bubble population spectrum, $\partial N/\partial R$, has an amplitude approximately 24 times the amplitude of the comparable freshwater spectral peak. The spectra obtained at various times after a "breaking wave event" have been used to estimate the temporal variation of the void fraction at the top of the plume, and from this estimate the speed of sound through the portion of a plume just below its associated whitecap, as a function of time, was ascertained, for both the freshwater and seawater cases.

Contributed Papers

11:15

2A08. Measurements of the low-frequency sound produced by 2-D and 3-D breaking waves. M. R. Loewen (R. M. Parsons Lab., 48-108, MIT, Cambridge, MA 02139) and W. K. Melville (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093 and R. M. Parsons Lab., 48-108, MIT, Cambridge, MA 02139)

Laboratory measurements of the sound beneath both 2-D and 3-D breaking waves are described. Sound at frequencies as low as 20 Hz was observed beneath the 2-D breakers and the mean-square acoustic pressure in the frequency band from 20 Hz–1 kHz correlated strongly with the wave slope and dissipation. The sound spectra observed beneath the 3-D breakers showed significant increases in level across the entire bandwidth from 10 Hz to 20 kHz and the spectra sloped at -5 to -6 dB per octave at frequencies greater than 1 kHz. The mean-square acoustic pressure in the frequency band from 10 to 150 Hz correlated with the wave amplitude similar to the results obtained in the two-dimensional breaking experiments. Large-amplitude low-frequency spectral peaks from 15 to 275 Hz were observed in the spectra of the sound from the 2-D and 3-D breaking waves. It is postulated that these low-frequency signals are due to the collective oscillation of the bubble clouds. Void fraction measurements were available for six of the breaking events (Lamarre and Melville, 1992, this meeting) and therefore the geometry and sound speed inside the bubble clouds were known. Using this information estimates of the resonant frequencies of the bubble clouds were made and these estimates were found to agree closely with the observed frequencies. The close agreement supports the hypothesis that the observed low-frequency signals were produced by the collective

oscillation of the bubble clouds. [Work supported by ONR (Ocean Acoustics).]

11:30

2A09. A laboratory experiment on nonlinear coupling between surface and volume modes of a bubble. Yi Mao, Lawrence A. Crum (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, University, MS 38677), and Ronald A. Roy (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

M. S. Longuet-Higgins has predicted [J. Fluid Mech. 201, 525–541 (1989)] that the distortion modes of a bubble can produce, through nonlinear coupling, a monopole radiation of sound. Here, some preliminary results are presented of an experiment designed to investigate this phenomenon. A hydrophone and needle are submerged in water in a small ($6.5 \times 6.5 \times 7.0$ cm³) sealed cell that is connected to a regulated evacuation system for pressure control. The sound radiation produced by releasing a bubble from the needle is monitored by a hydrophone and displayed on a digital oscilloscope. Two high-speed video cameras simultaneously record the motion of the bubble and the sound radiation trace as displayed on an analog oscilloscope. At low pressures (i.e., a few cm Hg), it is seen that after the pinch-off sound dies out, there follows a sound of the same frequency but with a lower and relatively constant magnitude, which we believe is evidence of the coupling between surface and volume modes. Observation of the phase between the surface oscillation and the radiated sound indicates a causal relationship. [Work supported by ONR.]

11:45–12:15

Panel Discussion

PANEL MODERATOR: Terry E. Ewart

PANEL MEMBERS: John E. Ffowcs-Williams

Michael S. Longuet-Higgins

Herman Medwin

W. K. Melville

Edward C. Monahan

Jeffrey A. Nystuen

Ming-Yang Su

Session 2EA

Engineering Acoustics: Transducers, Arrays, and Techniques

George S. K. Wong, Cochair

National Research Council, Institute for National Measurement Standards, Montreal Road, Ottawa, Ontario K1A 0R6, Canada

James M. Powers, Cochair

Naval Underwater Systems Center, Code 2131, New London, Connecticut 06320

Contributed Papers

8:00

2EA1. Open-circuit sensitivity of a PVF2 surface pressure sensor used for fetal heart tone monitoring. Allan J. Zuckerwar (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23665)

A portable fetal heart tone monitor has been developed using an array of PVF2 surface pressure sensors. The sensor array, mounted on a belt worn by the mother, detects pressure pulses from the fetal heart incident upon the maternal abdominal surface. Each sensor is constructed of components to fulfill five functions: signal detection, acceleration cancellation, acoustical isolation, electrical shielding, and electrical isolation of the mother. An analysis based on an acoustical model of the sensor yields the open-circuit sensitivity, which permits determination of the surface pressure levels. The sensor behaves like a low-pass filter having a cutoff frequency proportional to the stiffness of the belt material. Peak-to-peak surface pressures of the first (mitral-tricuspid) and second (aortic-pulmonary) heart sounds were determined to be 12.0 ± 3.6 Pa and 9.4 ± 2.8 Pa for a fetus of 34.5 weeks gestational age; and 47.0 ± 6.8 Pa and 27.9 ± 9.4 Pa for one of 39 weeks gestational age (full term).

8:15

2EA2. Active structural acoustic control with piezoelectric actuators and PVDF acoustic near-field sensors. Robert L. Clark, Jr. and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

Acoustic near-field sensors were designed from polyvinylidene fluoride (PVDF) and were implemented in the acoustic near field of a baffled simply supported plate as error sensors for active control of structure-borne sound. Sensor position was determined from nonlinear optimal design techniques as were the optimal positions of the piezoelectric control actuators. Both analytical and experimental results were obtained and compared to evaluate the sensor design as well as the control implementation for off-resonance excitation of the plate. Microphones were implemented as error sensors in the narrow-band cost function of the LMS control algorithm to provide a basis for comparison with the acoustic near-field sensors. The acoustic response of the structure was attenuated by 20 dB when implementing the acoustic near-field sensors, which was the same level of attenuation observed when implementing microphone error sensors in the radiated field. [Work supported by ONR/DARPA.]

8:30

2EA3. A dual frequency ultrasonic probe for medical application. Shiroh Saitoh and Mamoru Izumi (Electron Devices Lab., R&D Ctr., Toshiba Corp., Kawasaki, 210 Japan)

A dual frequency probe employing two ceramic layers for medical ultrasonic diagnostic equipment has been developed for simultaneously obtaining both a high-resolution tomogram and a high S/N Doppler mode image. This ceramic vibrator, which has opposite poling directions and different individual thicknesses, has enabled the excitement of a second harmonic in addition to the fundamental resonance. Therefore, only a pair of electrical terminals are enough to excite dual frequencies. Ultrasonic attenuation in a human body is reduced, since the Doppler reference frequency can be set below the tomogram center frequency. Moreover, the two relative resonant levels are widely controllable by changing the thickness ratio of the two layers. An optimum thickness ratio of 1:0.7 has been obtained by computer simulation under Mason's model. A cofiring multilayer method has been developed to achieve a multilayer ceramic with a large adhesive strength. As a result, the S/N of the Doppler mode has been improved as much as 5 dB over that of a conventional probe at an artery located 5 cm deep from the surface.

8:45

2EA4. New electrostrictive ceramics for sonar projector applications. Kurt M. Rittenmyer and R. Y. Ting (U. S. Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

Relaxor ferroelectric ceramics, in particular lead magnesium niobate-lead titanate ($\text{Pb}_3\text{MgNb}_2\text{O}_9\text{-PbTiO}_3$) compositions, have potential for hydroacoustic transducer applications. In an unbiased state, the strain-versus-electric field relationship is nonlinear (largely quadratic) with respect to the electric polarization. By applying a large dc bias voltage simultaneously with a small ac signal, acoustic signals whose amplitudes are linear with respect to the ac signal are produced. The strains produced are of similar magnitude to those of lead zirconate-titanate ceramics. Several specific compositions have been examined in order to optimize the material by reducing the dielectric and electromechanical hysteresis, the temperature dependence of dielectric properties, and the dielectric aging and the results are discussed. The possibility of being able to switch the polarization of the material off with the biasing field makes the further development of these materials intriguing. [This work supported by ONR.]

9:00

2EA5. Comparison of diffraction effects from an unfocused and focused circular transducer. Peng Jiang and Robert E. Apfel (Ctr. for Ultrasonics and Sonics, Yale Univ., P. O. Box 2159, Yale Sta., New Haven, CT 06520)

A simple theoretical model is used to study the diffraction effect of a pistonlike transmitter as well as a focused transmitter for receivers of different diameter in a transmission system used to measure the acoustic properties of a liquid, e.g., sound speed and attenuation. Numerical calculations show that, as one would expect, the diffraction effect from a focused transducer is smaller than that from an unfocused transmitter, and when the diameter of the receiver is large enough (more than twice that of the transmitter), the received pressure from a focused transmitter has approximately a plane wave character, e.g., constant amplitude and linear phase over a large range of the radiation field. The same cannot be said for an unfocused transducer due to phase problems. Results of experiments are presented in support of the theoretical study. [Work supported by the National Institutes of Health through Grant 5R01CA39374.]

9:15

2EA6. Composite fiber optic flexural disk hydrophones. D. Kopolka,^{*)} D. A. Brown, D. L. Gardner, and S. L. Garrett (Phys. Dept., Code PH/Br, Naval Postgraduate School, Monterey, CA 93943)

Fiber optic flexural disk hydrophones made from castable epoxy have previously been reported. These Michelson interferometric hydrophones produced excellent sensitivity and reasonable depth tolerance. They were designed in the shape of closed cylinders with fiber coils mounted on both sides of each end cap in a push-pull configuration. The current work seeks to increase the depth tolerance without sacrificing the sensitivity by constructing the end caps from composite materials. The present sensors were configured as Mach-Zehnder interferometers and terminated with 3×3 couplers. Three different composite materials were tested. The first was a quasi-isotropic lay-up of unidirectional E-glass in a thermoplastic matrix. The second was a 0°/45° lay-up of woven E-glass in a thermoplastic matrix. The third composite tested was a random orientation of E-glass fibers in a vinyl ester matrix. The theoretical sensitivity and depth tolerance of each composite sensor were calculated using the experimentally determined values of the dynamic Young's modulus, static flexural rigidity, and tensile strength and compared to measured results. [Work supported by NAVSEA.] ^{*)}Lt., USN.

9:30

2EA7. A prototype cylindrical wave-number calibration array. L. D. Luker (Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

There has been considerable interest in recent years in the acoustic interference of the turbulent boundary layer (TBL) produced on the surface of a vehicle moving through water. The "flow noise" produced by the TBL is generally characterized by pressure waves that are transported along the surface at speeds much less than the sound speed in water and are, therefore, evanescent. Planar array devices have been constructed previously that generate evanescent waves with desired spatial wave numbers in a region near the front surface of the array. This paper describes a prototype cylindrical wave-number calibrator array designed to generate the desired evanescent waves inside the free-flooded calibrator. This geometry is suitable for the evaluation of line acoustic sensors such as in towed sonar arrays. The prototype calibrator consists of a cylindrical tube of PVDF with an electrode pattern along the tube length that allows the PVDF to be operated as 40 independent bands each of which can be driven at a specified amplitude and phase. The required drive voltages are calculated by measuring the electroacoustic transfer matrix of the calibrator at the desired drive fre-

quency. Measured evanescent pressure fields inside the calibrator are compared with desired and predicted fields at selected spatial wave numbers for frequencies of 800 and 1600 Hz. [Work jointly supported by ONR and ONT.]

9:45

2EA8. Near-field array with constant focal area. Noel A. Adorno (Sony, Inc., Tokyo, Japan) and Elmer L. Hixson (Univ. of Texas, Austin, TX 78712)

A microphone array to be placed in the headliner of an automobile was designed for Cellular telephone application. The design goal was to provide a focal area of 20×20 cm centered at the driver's mouth to be held constant from 300 to 3000 Hz. Digital filtering-delay techniques were used to control focal area. Array size limitations and a limited number of elements produced focal area sizes of 19×19 to 24×24 cm from 450 to 3000 Hz.

10:00

2EA9. Constant directivity receiving arrays. J. Michael Williams, Michael D. Cerna (Natl. Instruments, Austin, TX 78730-5039), and Elmer L. Hixson (Univ. of Texas, Austin, TX 78712)

By combining a linear array with its half-size model, through specified filters, constant mainlobe directivity over an octave frequency range can be achieved. Digital implementation of the filter is particularly useful to provide time delays to endfire arrays and filter functions that cannot be implemented with analog circuits. Additional half-size arrays and filters can be used to extend the bandwidth over several octaves. Results for broadside and endfire arrays are presented.

10:15

2EA10. Experimental determination of the three-port impedance matrix for an array of transflexural elements. Daniel M. Warren, L. Dwyann Lafleur, and F. Douglas Shields (Natl. Ctr. for Phys. Acoust., Coliseum Rd., University, MS 38677)

A transducer has been built that consists of an array of transflexural piezoelectric disks embedded in a soft potting material. This transducer can be configured as a thin layer yet exhibits significant low-frequency (audio band) sensitivity. It has been modeled as a general three-port transducer, and an experimental determination of the three-port impedance matrix over a range of frequencies for a prototype transducer has been made. The matrix is expressed in terms of effective constants (the electromechanical coupling constant, the clamped dielectric constant, and the elasticity constant) of a thickness mode piezoelectric transducer. Predictions of the frequency response of the transducer and its performance as an active surface have been made using the measured impedance matrix. These predictions have been compared to its actual performance as an active surface.

10:30

2EA11. Development and comparisons of two cylindrical passive arrays. Thomas R. Howarth, Mark L. Pecoraro, Alan K. Walden, and Allan C. Tims (Naval Res. Lab., Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

Two cylindrical, high-gain, wide-band (three octaves) passive arrays have been recently developed. Both arrays consist of 36 lead titanate elements mounted on a cylindrical structure. The elements are arranged in a 3×12 configuration with the 12 elements placed along 180° of the cylinder's circumference. The difference between the design of the two arrays is in the backing of the individual piezoceramic elements. The elements in the first array have a steel tail mass behind them to

form an acoustic hard boundary while the elements of the second array have a coprene layer backing to form an acoustic soft boundary. Other differences are that the first array has its elements in an oil-filled boot and the second array is potted in polyurethane. The presentation shall include conceptual drawings of the configuration of the elements and the arrays. Performance comparisons shall include the measured free field sensitivity, acoustic radiation patterns, and equivalent self-noise. [Work sponsored by DTRC.]

10:45

2EA12. Constant directivity loudspeaker arrays. Jefferson A. Harrell (Jet Propulsion Labs., Pasadena, CA 91109) and Elmer L. Hixson (Univ. of Texas, Austin, TX 78712)

It is shown that the beamwidth of the major lobe of loudspeaker arrays can be held constant over wide frequency ranges by suitably combining superposed arrays. Each array is excited through specified digitally implemented filters. Two-dimensional coverage is achieved with broadside and endfire arrays over an octave bandwidth. The method is particularly useful at low frequencies where constant directivity horns are ineffectual.

11:00

2EA13. Performance of various hydrophone shapes in the reduction of turbulent flow noise. Sung H. Ko (Naval Undersea Warfare Ctr. Detachment, New London, CT 06320)

Turbulent boundary layer pressure fluctuations can be reduced by either spatial filtering through a finite hydrophone or a hydrophone array or by filtering through an elastomer layer. In general practice, various configurations of hydrophone arrays are embedded within a layer of elastomer, thus reducing the turbulent boundary layer pressure fluctuations. The theoretical model considered in this paper is a plane elastomer layer backed by an infinitely rigid surface; the other side of the layer is exposed to turbulent flow. This paper examines the performance of various shapes (square, rectangle, triangle, and rhombus) of hydrophones embedded within a layer of elastomer in reducing the turbulent flow noise. The results presented are numerically calculated noise reductions for various parameters related to hydrophone shapes.

11:15

2EA14. Theory of iterative time reversal acoustic mirrors. C. Prada, J. L. Thomas, and M. Fink (LOA, ESPCI, 10 rue Vauquelin, 75005 Paris, France)

A new ultrasonic focusing method for reflective target has been demonstrated in the laboratory. This method is interesting for nondestructive testing and lithotripsy. It uses an ultrasonic time reversal mirror made of a 2-D array of transmit and receive transducers each connected to a programmable generator. The array first transmits a wave that is reflected by the target. The reflected wave is received by the same array and then the corresponding signals are stored, time reversed, and

re-emitted. The resulting wave focuses on the target even when the propagating medium is inhomogeneous. This process can be iterated and when the medium contains several reflectors that are well resolved, this allows one to focus on the strongest one. These self-focusing properties have been demonstrated theoretically and experimentally [C. Prada *et al.*, *J. Acoust. Soc. Am.* **90**, 1119–1129 (1990)]. In the general case, the convergence of the iterative process is not obvious. A matrix theory, valid for an inhomogeneous propagating medium and reflectors of complex scattering properties, shows that the process converges but may yield to different limits for odd and even numbers of iterations. This analysis is illustrated by several numerical and experimental examples.

11:30

2EA15. A new design for ultrasonic wedge transducer. Zhi Hua Shen (Analogic Scientific, Inc., 6th Industrial Rd., SheKou, ShenZhen, Guang Dong, People's Republic of China) and Dehua Huang (Dept. of Phys. Acoust., Univ. of Mississippi, University, MS 38677)

The ultrasonic wedge transducer has attracted much interest in recent years in the fields of microscopy, medical ultrasonic, and nondestructive testing evaluation. The new design gives a better particle velocity distribution on the surface of the wedge transducer. When the Gaussian transducer technique is involved in the design, a Gaussian wedge transducer is achieved. The theory behind the design is also presented.

11:45

2EA16. A three-parameter model for the behavior of transducers. Li-Feng Ge (Anhui Bureau of Standards and Metrology, Hefei, Anhui 230001, People's Republic of China)

Owing to the limitation of materials and technology, it is significant to determine the behavior of a transducer as a whole by experiments. The impedance method is one of the most important methods. In previous works, a four-parameter model was established to relate the mechanical load impedance to its input electrical impedance. The characterized parameters of the transducer were then determined from the model parameters. [L.-F. Ge, *J. Acoust. Soc. Am. Suppl.* **1** **87**, S129 (1990); *J. Acoust. Soc. Am.* **89**, 1860 (1991)]. But, results obtained by the model vary more or less with different beginning values calculated even from proper three-pair impedance data. In this paper, a three-parameter model is presented, and a complex least-squares fitting code is developed to fit the model to more than three pairs of data; thereby, a definite and more accurate solution has been obtained. The model is formulated as $Z_E = A^*(Z_L - B^*) / (Z_L - C^*)$, where Z_E is the electrical impedance, Z_L is the load impedance, and the multiplier A^* , pole point C^* , and zero point B^* are the three parameters to be determined. Exactly, A^* is equal to the mechanically blocked electrical impedance and C^* is equal and opposite to the electrically blocked mechanical impedance. The new model has been successfully used to determine the characterized parameters, motional impedance, and sensitivity of both piezoelectric and electrodynamic vibration generators, and then the sensitivity of accelerometers.

Session 2NS

Noise: Governmental Approaches to Noise Control

Alice H. Suter, Chair

Alice Suter and Associates, 1657 River Dee Court, Cincinnati, Ohio 45230

Chair's Introduction—8:30

Invited Papers

8:35

2NS1. Recent European legislation in the field of noise abatement and its impact on standardization. Klaus Brinkmann (Physikalisch-Technische Bundesanstalt, Bundesallee 100, D-3300 Braunschweig, Germany)

According to the "new approach" from 1985, European legislation in the field of technical harmonization is limited to the adoption of essential safety requirements, while the task of drawing up related technical product specifications is entrusted to European standardization organizations. The "global approach" from 1989 describes the policy of the European Communities regarding conformity assessment procedures. Noise aspects are covered, for example, by the Directives 86/188/EEC on the protection of workers against noise and 89/392/EEC on machinery safety. The latter is of the "new approach" type and contains essential requirements with respect to the measurement, declaration, and reduction of noise. It has initiated considerable standardization activity, both on the European level (e.g., within CEN/TC 211, Acoustics) and the international level (e.g., within ISO/TC 43, Acoustics, and its Subcommittee 1, Noise). The cooperation established between these committees is completely in accordance with the recent Vienna Agreement between CEN and ISO. The procedure and the present status of this common standardization effort in the field of acoustics will be described.

9:00

2NS2. The dormant Noise Control Act and options to abate noise pollution. Sidney A. Shapiro (School of Law, Univ. of Kansas, Green Hall, Lawrence, KS 66045)

In 1981, Congress ended funding for the Office of Noise Abatement and Control (ONAC) at the Environmental Protection Agency (EPA). Before the elimination of ONAC, EPA engaged in a wide variety of activities to abate noise pollution under the authority of the Noise Control Act and, after 1978, the Quiet Communities Act. Elimination of ONAC's funding has stymied noise abatement efforts at both the federal and state level, while noise levels have probably increased. Without federal action, little governmental noise abatement activity can be expected. Although Congress has several options to abate noise pollution, funding EPA to implement the NCA is the best solution.

9:25

2NS3. Federal review of the aircraft noise assessment process. William D. Dickerson (U.S. Environmental Protection Agency (A-104), 401 M St. SW, Washington, DC 20460)

The Federal Interagency Committee on Noise (FICON) was formed in December 1990 as an outgrowth of airport noise analysis concerns raised by the Environmental Protection Agency. The committee's function is to review the Federal approach for airport noise assessment used in Environmental Impact Statements (EIS). The committee has made preliminary recommendations for improving the airport noise assessment process and has circulated their report for public comment. The recommendations cover the continued use of the DNL metric, use of supplemental metrics, how these metrics are described in documents available to the affected public, the scope of analysis in EISs, mitigation options used in EISs, and the need for continued systematic public input to improving the process. After public review, final recommendations will be made to Federal Agency decision makers for their consideration.

Break

10:00

2NS4. Local governmental strategies for noise control. Frank C. Gomez (Environmental Health, Dept. of Health Services, County of Los Angeles, 2525 Corporate Pl., Monterey Park, CA 91754)

Since the demise of the EPA Noise Control Program in 1982, local and state noise programs have had a difficult time developing the expertise to control noise. In California, state and local support for community noise control programs have been reduced while the public demand for noise control has increased. In an effort to provide support to cities that need to develop noise control programs several acoustical consultants and local governmental noise officials have sponsored statewide training programs. Local noise control training is required if local government is to have an effective and enforceable noise control program. Furthermore, local noise ordinances must be easily understood, use a simplified monitoring metric such as a L_{eq} standard, and use equipment which can be properly operated by staff.

10:25

2NS5. The role of voluntary standards in a National Noise Control Program. Kenneth McK. Eldred (Ken Eldred Engineering, P.O. Box 1037, Concord, MA 01742)

Voluntary standards can and should play a central role in a National Noise Control Program. They provide a uniform vocabulary for noise and its effects on people both in the USA and throughout the world as a result of efforts to harmonize national and international standards. They contain methods for the objective measurement of the noise characteristics of sources, evaluation of sound propagation effects, and characterization of the spatial and temporal aspects of environmental noise. Voluntary standards in acoustics in the USA are developed by several organizations, including the Acoustical Society of America, the Society of Automotive Engineers, and the American Society of Testing and Materials. They are the product of concerned professional individuals and representatives of directly concerned interests including both government and industry as well as organizations representing concerned technical and public groups.

Contributed Papers

10:50

2NS6. Comparison of the speech transmission index and the modified rhyme test in simulated cockpit ambient noise. Lisa A. Griffin (Northrop Corp./B-2 Div., 6510 Test Squadron/ENFH, Mail Code EW010/5D, Edwards AFB, CA 93523)

The military aircraft test and evaluation community has recently introduced a much quicker, less obtrusive method for evaluating speech intelligibility of a communication system. The method uses the Speech Transmission Index (STI) to predict intelligibility using 15 s of flight time. Previous research has developed a correlation between this method and subjective intelligibility scores for the Phonetically Balanced (PB) word test. This research has not yet shown an experimental relationship to the Modified Rhyme Test (MRT), one preferred method of the military. In addition, the research has not accounted for ambient noise with a spectrum that represents an in-flight aircraft cockpit environment. In the present study, the relationship between the STI and MRT in simulated noise was investigated. Two types of noise were used. The first type was noise with a spectrum to represent in-flight ambient cockpit noise. The second was pink noise (equal energy per octave), a commonly used representation of environmental cockpit noise. The study was two phase. Phase one measured the intelligibility of a communication channel degraded with the noise presented at different signal-to-noise ratios. The intelligibility test was the MRT. Phase two used the same communication channel and signal-to-noise ratio conditions to measure objectively the intelligibility using the STI. The findings of the study showed that the MRT scores were significantly differ

ent between the different noise presentations while the MRT scores and STI yielded approximately the same values.

11:10

2NS7. Effects of noise on crewmembers during Spacelab Life Sciences-1. Anton S. Koros, Charles D. Wheelwright (Lockheed Eng. & Sci. Co., Mail Code C-95, 2400 NASA Rd. 1, Houston, TX 77058), and Susan Adam (Johnson Space Ctr., Mail Code SP 34, Houston, TX 77058)

Excessive noise can lead to decrements in performance, impaired verbal communication, fatigue, and hearing damage. On the STS-40/SLS-1 mission noise levels were evaluated through the use of a questionnaire and two objective measures—in-flight sound level measurements and pre- and post-flight crew audiometry tests. Sound level meter measurements suggested that noise levels in Spacelab during nominal operations were approximately 70 dB (A-weighted). This is in excess of the current acoustic specification of 59 dB (A-weighted). Noise measurements conducted in the Orbiter did not exceed current Shuttle standards. Crewmembers recommended that noise levels be reduced. Post-flight audiometry tests indicated that transient decreases in hearing ability had taken place during the mission. The average decrease in hearing level of 4.34 dB was statistically significant. Crewmembers noted that sleep, concentration, and relaxation were negatively impacted by high noise levels. Communication was also hampered. The higher than desirable noise levels in Spacelab were attributed to flight specific payloads for which acoustic waivers were granted. It is recommended that current noise levels be reduced in Spacelab and the Orbiter

Middeck. Levels of NC 50 are recommended in areas where speech communication is required, and NC 40 in sleep areas—in accordance with current Space Station Freedom standards. [This research was supported by the National Aeronautics and Space Administration under Contract No. NAS9-17900.]

11:50

2NS9. Acoustics of a high rise residential project. Ballard W. George (Earth Metrics, Inc., 7000 Marina Blvd., 4th Fl., Brisbane, CA 94005)

This paper describes acoustical elements of an urban residential project from design stage through construction. The project consists of nine stories of residential units over commercial space and occupies a city block in San Francisco. Design objectives included achieving consistency with, and in some cases bettering, California Title 24 Noise Insulation Standards. These standards are concerned with control of exterior to interior noise and the sound insulation afforded by common partitions. Other topics addressed included: mechanical equipment; plumbing systems; ceiling treatment; elevators; trash chutes; fire safing material; fire alarm horns; interconnecting ducting; duct attachments to structure; garage door mechanism; sprinkler pipe; and unit entry doors near commercial space. Fire safing effectiveness for acoustics depends on both resiliency and transmission loss; sometimes tradeoffs are needed. An investigation was made of the feasibility of placing fire alarm horns in the corridors; however, in the final design fire alarms were located in each dwelling unit for best audibility. Comments were received from a neighbor concerning an audible tone from rooftop equipment; an investigation was made to locate the specific source, and findings are reported.

11:30

2NS8. U. S. Navy shipboard noise criteria. Louis A. Herstein, III (Analysis & Appl. Res. Div., Tracor Appl. Sci., Inc., Ste. 1100, 2361 S. Jefferson Davis Hwy., Arlington, VA 22202-3863) and David D. Bernhard (Naval Sea Systems Command, Arlington, VA 20362-5101)

Steady state airborne noise criteria for communication and habitability within shipboard spaces have evolved from simple qualitative statements, through octave-band limits, to A-weighted levels. It is now recommended that A-weighted criteria supplemented by C-weighted minus A-weighted criteria be used to avoid the occurrence of low frequency (less than 200 Hz) problems. The recommended levels, while still somewhat higher than those considered acceptable for shore based environments because of ship cost, weight, and space restraints, should result in improved shipboard concentration and habitability. Octave band criteria related to the A-weighted and C-weighted minus A-weighted criteria would continue to be used for engineering design purposes. This paper will briefly describe the development of the criteria. [Work supported by Naval Sea Systems Command.]

TUESDAY MORNING, 12 MAY 1992

SALON B, 8:15 TO 11:50 A.M.

Session 2PA

Physical Acoustics: Nonlinear Physical Systems I

Robert M. Keolian, Chair

Department of Physics, Code PH/Kn, Naval Postgraduate School, Monterey, California 93943

Chair's Introduction—8:15

Invited Papers

8:20

2PA1. Toward linear macrosonics. Timothy S. Lucas (Sonic Compressor Systems, Inc., 3642 Chauncey Court East Dr., Lafayette, IN 47905)

Large resonant acoustic pressure amplitudes were mechanically generated inside resonators filled with refrigerant HFC-134a. The nonlinear losses normally associated with resonant macrosonic waves were dramatically attenuated via resonators whose higher resonant modes were noninteger multiples of the fundamental. Such resonators reduced Q amplification of the fundamental's harmonics, minimized viscous and thermal wall losses and minimized peak acoustic velocity. These noninteger resonators provided acoustic pressure amplitudes of up to 60 psi peak-to-peak for ambient pressures of 70 psi. At peak acoustic pressure amplitudes of 25% ambient, measured acoustic energy dissipation was within 34% agreement with strictly linear predictions. A high-amplitude resonator was used to construct a laboratory prototype of an oil-free resonant acoustic compressor. Compression ratios were provided via high-speed valves that converted acoustic pressure oscillations into an external discharge pressure. A 300-Hz discharge valve was tested that converted 95% of a resonator's peak acoustic pressure amplitude into an external discharge pressure. Efficiency calculations indicate that resonant acoustic compressors could provide improved efficiency for home refrigerators. [Work conducted at Los Alamos National Laboratory in collaboration with Greg Swift and supported by DOE.]

2PA2. Kink-assisted mode hopping in a resonator. Bruce Denardo, Andrés Larraza, and Charles McClelland (Dept. of Phys., Code PH/De, Naval Postgraduate School, Monterey, CA 93943)

For a sufficiently large fixed drive amplitude, a localized kink structure is observed to spontaneously participate in the transition of one standing wave mode to another as the drive frequency is slowly changed. The observations are of gravity waves in a parametrically driven annular channel of liquid. The kink, which can occur at any location around the resonator, has a substantially greater amplitude and shorter wavelength than the extended mode region. Such a structure is predicted to exist according to a theory that simultaneously allows amplitude *and* wave-number modulations of a finite-amplitude standing wave. The situation is in fundamental contrast to nonlinear Schrödinger (NLS) solitons, which correspond to only amplitude modulations, and to all other known types of solitons. Observations and theory of the NLS structures will be reviewed, and their role in the discovery of the new kinks will be discussed. The possibility of propagating versions of the new kinks, and of applications to fiber optic communications, will also be discussed.

Contributed Papers

9:20

2PA3. Absorption of sound by noise in one dimension. Andrés Larraza and Bruce Denardo (Dept. of Phys., Code PH/La, Naval Postgraduate School, Monterey, CA 93943)

The problem of the statistical closure for one-dimensional interactions of nondispersive random waves is apparently ill posed. The problem consequently lacks an H-theorem formulation for how the system approaches thermodynamic equilibrium. Rudenko and Chirkin have, on the other hand, successfully computed the attenuation of an infinitesimal coherent wave due to the nonlinear interaction with broadband noise in one dimension [O. V. Rudenko and A. S. Chirkin, *Sov. Phys. JETP* **40**, 945–949 (1975)]. They have shown that the attenuation coefficient is a linear function of position. This remarkable result implies that in one-dimensional acoustics, translational invariance is broken, the fluctuating forces are nonlocal, and the system is necessarily far from equilibrium. The physical implications of this result, and the connections to the fluctuation-dissipation theorem, will be discussed.

9:35

2PA4. Nonlinear attenuation of 10- to 25-MHz ultrasonic waves in argon at pressures up to 10 MPa. L. J. Bond, B. Dube, C-H. Chiang, C. M. Fortunko, and J. D. McColskey (Materials Reliability Div., Natl. Inst. of Stand. and Technol., 325 Broadway, Boulder, CO 80303-3328)

A large dynamic range (70 dB) ultrasonic measurement system has been developed to investigate the technology and the phenomenology of finite-amplitude high-frequency (10–25 MHz) acoustic waves in gases at pressures up to 10 MPa (100 bar). Transmission measurements of velocity and attenuation are made using two efficient lithium niobate transducers set onto quartz buffer rods, with variable separation from less than 1 to 25 mm. In this paper, data will be presented that show nonlinear phenomena as a gated tone burst (20 cycles) of finite amplitude acoustic waves propagated in argon. The received signals are digitized and analysis is performed in the time domain and using a Fourier transform. For low powers, linear attenuation with distance is observed. For finite amplitude waves, excess nonlinear attenuation and significant second harmonic generation are observed. The nonlinearity parameter at various combinations of pressure, temperature, and frequency is reported for argon.

9:50

2PA5. Finite amplitude standing waves in harmonic and anharmonic closed tubes. D. Felipe Gaitan and Anthony A. Atchley (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

Recent developments in thermoacoustic devices have generated a renewed interest in finite amplitude standing waves in resonant cavities. Such devices can generate standing waves with acoustic pressure amplitudes on the order of 10% of ambient pressure. The similarity between previously observed finite amplitude waveforms in closed tubes and those in prime movers could be modeled as anharmonic resonating cavities. Measurements of the energy transfer into the harmonics and the resonant modes of both harmonic and anharmonic closed tubes driven at resonance (~ 200 Hz) will be presented. These measurements were compared with calculations using Coppens and Sanders' formulation [A. B. Coppens and J. V. Sanders, "Finite-amplitude standing waves within real cavities," *J. Acoust. Soc. Am.* **58**, 1133–1140 (1975)], which requires the measured quality factors and resonance frequencies. [Work supported by ONR and the NPS Research Program.]

10:05

2PA6. Water fountains produced by intense standing waves in air. Thomas W. VanDoren and Mark F. Hamilton (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712-1063)

An acoustic demonstration found in several science museums involves a horizontal standing wave tube that contains a shallow layer of liquid. When the liquid is water and the sound pressure level of a standing wave in the air above the water exceeds approximately 160 dB (*re*: 20 μ Pa) at frequencies below 1 kHz, vigorous fountains occur at the nodes in the pressure field. The introduction of a surfactant in the water causes clusters of fountains to appear near the nodal planes at somewhat lower sound pressure levels. Standard quasilinear theory provides a reasonable description of the acoustic streaming and dc pressure in the air prior to the appearance of fountains. The fountains seem to result from the ejection of droplets by inner streaming vortices that are formed above the air–water interface. The water fountains and streaming patterns are discussed in relation to Andrade's observations [*Proc. R. Soc. London Ser. A* **230**, 413–445 (1932)] of solid particulate motion produced by intense standing waves in air. [Work supported by a Rockwell Graduate Fellowship, the ONR, and the Packard Foundation.]

Break

10:35

2PA7. Turbulent energy spectrum predictions by nonlinear acoustic scattering. James E. Parker, III and Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402)

An underwater nonlinear experiment is performed involving the propagation of finite-amplitude sound waves through a turbulent flow. The experimental geometry involves the interaction of two mutually perpendicular crossed ultrasonic beams that are generated by individual focused transducer units (with frequencies of 2.0 and 2.1 MHz, respectively). The alignment positions the center of the interaction at the overlapping focal regions. A mechanical apparatus allows the crossed beams to rotate in a horizontal plane containing the submerged circular water jet. The beam axes always remain perpendicular. In the presence of turbulence, a radiated nonlinear sum frequency component (f_+ = 4.1 MHz) is scattered and detected by a receiving transducer unit located outside the interaction region. In the absence of turbulence, there is virtually no nonlinear scattering. Measurements of the broadened intensity spectrum versus angle contain statistical information about the turbulent velocity components. Further analysis shows that focused crossed beam scattering exhibits good spatial resolution. From theory [Korman and Beyer, *J. Acoust. Soc. Am.* **85**, 611-620 (1989)] the energy spectrum of turbulence $E(k,r)$ (where k is the wave number of the turbulent eddies) is predicted by performing angular measurements at several scan positions r across the jet. [Work supported by the Naval Academy Research Council.]

10:50

2PA8. Observation of single-point excitation and shocking of rotating acoustic waves. Peter H. Ceperley and Alon Koren (Depts. of Phys. and Elec. and Comput. Eng., George Mason Univ., Fairfax, VA 22030)

In a cylindrically symmetric resonator or a ring resonator, the proper combination of standing wave modes results in a rotating wave field. Such a field has the appearance of a traveling wave chasing its tail, a constant field profile rotating in space. However, unlike traveling waves, rotating waves are limited to distinct modes. Rotating waves offer the clearest insight into angular momentum and rotary motion in wave fields. The following observations will be reported: (1) rotating acoustic waves in a concentric ring resonator and in a cylindrical resonator; (2) single-point excitation of these waves [P. H. Ceperley, "Split mode traveling wave ring resonator," U.S. Patent 4,686,407 (1987)]; and (3) weak shocking of the wave fronts at acoustic amplitudes near 1 kPa. These all relate to one atmosphere, air-filled resonators. [Work supported by ONR.]

11:05

2PA9. Further investigation of a single, stable, sonoluminescing bubble using Mie scattering. D. Felipe Gaitan, W. Jerry Lentz, and Anthony A. Atchley (Phys. Dept., Naval Postgraduate School, Monterey, CA 93943)

A year after its discovery, a single sonoluminescing bubble is far from being understood. The role of the various physical parameters such as vapor pressure, ambient temperature and pressure, etc., are yet to be determined. To this end, we have varied these parameters while monitoring the dynamic radius of the bubble using Mie scattering. Furthermore, the nature of the sonoluminescence emissions has been investigated by measuring the overall intensity as well as spectra as a function of these parameters. [Work supported by ONR and the NPS Research Program.]

11:20

2PA10. Limitations of the hydrodynamical theory of cavitation induced sonoluminescence. Ritva Löfstedt, B. P. Barber, R. Hiller, and S. Putterman (Phys. Dept., Univ. of California, Los Angeles, CA 90024)

The extremely short duration (< 50 ps) of the flashes of light emitted by a sound field raises problems for the hydrodynamical description of sonoluminescence. Solutions to the Navier-Stokes equations with such a fast but short-lived adiabatic compression develop shock fronts. Even in the linear approximation a single $20\text{-}\mu$ bubble that accelerates at this rate would radiate so much sound that its motion would dominate the Q of a 0.1-liter resonator. The extent to which such a quickly accelerating bubble can be considered to be in local equilibrium is discussed. Nevertheless, fluid mechanics provides valuable scaling laws that can be used to analytically determine the ambient radius, phase of collapse, and maximum and minimum radii of the trapped bubble's oscillation. [Work supported by the USDOE: Office of Basic Science (theory) and Division of Advanced Energy Projects (experiment) and R. L. is an AT&T fellow.]

11:35

2PA11. A mechanism for the rapid quenching of sonoluminescence pulses. Andrea Prosperetti and Hasan Oğuz (Dept. of Mech. Eng., The Johns Hopkins Univ., Baltimore, MD 21218)

It has recently been found that the duration of sonoluminescent light emission from stably oscillating bubbles is of the order of 200 ps or shorter [Barber and Putterman, *Nature* **352**, 318 (1991)]. This result is very surprising because, according to the best available estimates, temperatures high enough to cause luminescence from excited OH radicals persist for times 3-4 orders of magnitude longer [Egolfopoulos *et al.*, *J. Acoust. Soc. Am.* (submitted)]. A possible mechanism to explain the observed premature quenching of the light emission is as follows. Near the time of minimum radius a very strong acceleration exists directed from the gas into the liquid. This situation is Rayleigh-Taylor unstable and may cause the shedding of minute liquid droplets from the bubble surface into the gas. These drops may cool the gas sufficiently to prevent further light emission. To investigate this mechanism a boundary element calculation of the evolution of the surface shape of a collapsing bubble is performed. [Work supported by NSF.]

Session 2PP

Psychological and Physiological Acoustics: Temporal Processing, Modulation, and Source Separation

William A. Yost, Chair

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

Chair's Introduction—8:25

Invited Paper

8:30

2PP1. On the detection and perception of modulation by human listeners. William Morris Hartmann (Dept. of Phys., Michigan State Univ., East Lansing, MI 48824)

In a series of papers, beginning in 1952, Eberhard Zwicker and his colleagues redefined the psychoacoustics of modulation detection and perception. The scope of this subject now encompasses questions of considerable generality: whether the detection of modulation is effectively a measure of difference limens, or whether the temporal variation present in the modulation introduces specific elements of neural processing; whether frequency modulation detection and amplitude modulation detection can both be understood from a single excitation-pattern model, and, whether auditory temporal processing of all kinds can be understood in terms of superposition of responses to modulated tones. Presently, there appears to be a totally satisfactory understanding of modulation detection at high modulation rates in terms of a spectral model and tone-on-tone masking. However, attempts to find a unified model for low modulation rates, where temporal processing is indicated, must cope with some difficult experimental results, including data that show awkward dependences of modulation detection and difference limens on signal parameters, evidence of selective adaptation, and recently discovered cross-channel effects. [Work supported by the NIDCD, DC00181.]

Contributed Papers

9:00

2PP2. Detection of combined frequency and amplitude modulation. Brian C. J. Moore (Dept. of Exptl. Psychol., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, England) and Aleksander Sek (Inst. Acoust., Adam Mickiewicz Univ., 60-769 Poznan, Poland)

Zwicker [Acustica 6, 356–381 (1956)] proposed that amplitude modulation (AM) and frequency modulation (FM) are coded by a common mechanism. To test this, the detection of simultaneously occurring AM and FM is discussed. In a two-alternative forced-choice task, thresholds for detecting AM alone were determined. Then, thresholds for detecting FM were determined for stimuli which had a fixed amount of AM in the signal interval only. The amount of AM was always less than the threshold for detecting AM alone. The FM thresholds depended significantly on the magnitude of the coexisting AM. For low modulation rates (4, 16, and 64 Hz), the FM thresholds did not depend on the relative phase of modulation for the FM and AM. For a high modulation rate (256 Hz) strong effects of modulator phase were observed, which can be explained by assuming that detection of modulation at high frequencies is based on detection of the lower sideband in the modulated signal's spectrum. In a second experiment, psychometric functions were measured for the detection of AM alone and FM alone. For each type of modulation, d' was approximately a linear function of the square of the modulation index. Application of this finding to the results of experiment 1 suggested that, at low modulation rates, FM and AM are not detected by completely independent mechanisms.

9:15

2PP3. Modulation masking in normal-hearing listeners and in auditory brainstem implant patients. Chris Ahlstrom and Robert V. Shannon (House Ear Inst., 2100 West Third St., Los Angeles, CA 90057)

Recent experiments have suggested that a modulated masker can interfere with detection of a signal of similar modulation rate, in a manner analogous to the familiar critical-band concept in the frequency domain [Houtgast, J. Acoust. Soc. Am. 85, 1676–1680 (1989)]. An experiment with sinusoidally modulated maskers did not show such a critical-band effect in either normal-hearing listeners or in patients electrically stimulated on the cochlear nucleus. Maskers consisted of 500- or 1000-Hz sinusoidal carriers sinusoidally modulated at 4, 10, or 16 Hz with a modulation index of $m = 0.5$. Thresholds for modulation depth were obtained for the same carrier modulated at signal frequencies of 1 to 32 Hz (in 1-Hz steps). Modulation masking was observed only when the masker and signal modulation frequencies were equal. The magnitude and form of the masking patterns depended strongly on the relative phase of the two modulators. Prior results might have shown critical-band type interference patterns because of the stochastic nature of the modulated maskers used. Previous and present results can be explained adequately by a short-term envelope correlation, without need to postulate a central mechanism for grouping temporal modulation patterns in the manner of a filter. [Work supported by NIH.]

9:30

2PP4. Spectral transposition of envelope modulation. Stanley Sheft and William A. Yost (Parmly Hear. Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

Previous work [Sheft and Yost, *J. Acoust. Soc. Am. Suppl.* 1 **85**, S121 (1989)] has shown that it is often difficult to identify which component of a multicomponent complex is amplitude modulated. The role of the carrier in envelope processing was examined in the present study with a cued 2IFC envelope-discrimination procedure. The narrow-band noises of the two observation intervals differed only in terms of their pattern of envelope fluctuation, with the signal interval a repetition of the cue envelope pattern at a center frequency (CF) different from that of the cue. Cross-spectral transposition of envelope information was evaluated by varying the number of common CFs between the noise bands of the cue complex and the target bands of the observation intervals. With both the cue and observation intervals consisting of a single noise band, the ability to transpose envelope information from the cue to the observation-interval CF diminished with increasing noise bandwidth from 12.5 to 200 Hz. Results from the multiband conditions indicate that listeners are unable to integrate the envelope information across audio-frequency regions to improve performance. In fact, the added noise bands led to a significant drop in performance in many conditions. These results suggest that envelope information is not processed inde-

pendent of the spectral location of the modulated carrier. [Work supported by NSF and AFOSR.]

9:45

2PP5. Discrimination of tonal sequences composed of repeated temporal patterns. T. Sadralodabai and R. D. Sorkin (Dept. of Psychol., Univ. of Florida, Gainesville, FL 32611)

The discrimination of rhythmic and nonrhythmic temporal patterns was investigated. Two sequences of 12 tones were presented successively (tone duration = 25 ms, frequency = 1000 Hz). The temporal pattern of each sequence was defined by the time intervals between the tones. The task was to report whether the two sequences had the same or different temporal patterns. According to the pattern correlation model [R. D. Sorkin, *J. Acoust. Soc. Am.* **87**, 1695-1701 (1990)], listeners perform the task by computing the correlation between the two temporal patterns. In the current experiment, two aspects of these sequences were manipulated: (a) the correlation between the two 12-tone temporal patterns, and (b) the rhythmicity of the sequences, which we define as the correlation between the temporal patterns of successive 4-tone subsequences within a 12-tone sequence. Listeners' performance increased with the rhythmicity of the sequences. Performance was above the level predicted by a simple extension of the pattern correlation model [Work supported by AFOSR.]

10:00-10:15

Break

10:15

2PP6. Detection of mistuning in the presence of an interfering sound. Robert P. Carlyon (Exptl. Psychol., Sussex Univ., Brighton BN1 9QG, England)

Mistuning is an important cue for the perceptual segregation of concurrent sounds. Although it often has to be detected when more than one sound is present, the effect of extraneous sounds on sensitivity to mistuning remains unknown. This was investigated using a 2AFC task, where the standard consisted of the first seven harmonics of 500 Hz, all frequency modulated coherently (5-Hz rate, 200-ms duration) by the same percentage of their starting frequencies. The initial modulation phase was randomized across presentations. The signal was identical, except that the fourth harmonic was modulated in antiphase *re*: the other components, causing it to become mistuned by an amount proportional to the FM depth (the dependent variable). Stimuli were presented at 45 dB SPL per component in a continuous pink noise. The threshold FM depth necessary for the detection of mistuning was measured as a function of the level of a 2000-Hz unmodulated sinusoid. When this "interferer" was gated on and off with the complex, it had only a small effect on thresholds provided that its level was lower than about 42 dB SPL. However, when it was gated on 400 ms before, and off 100 ms after, the complex, thresholds were *higher* than with a synchronous interferer, and were elevated even at low interferer levels. Continuous interferers produced thresholds intermediate between the two conditions. The effect of interferer asynchrony persisted when the interfering sinusoid was replaced by narrow-band noise and is interpreted in terms of auditory streaming.

10:30

2PP7. Auditory stream segregation by musical timbre. Paul Iverson (Dept. of Psychol., Uris Hall, Cornell Univ., Ithaca, NY 14853-7601)

A series of experiments examined the acoustic attributes of timbre that most contribute to auditory stream segregation. In the first experiment, listeners heard short sequences of repeated pairs of tones from orchestral instruments, and were asked to rate how strongly the tones formed different streams. These ratings were used as difference measures, and were fit to a two-dimensional space using multidimensional scaling. One dimension corresponded to average spectral frequency, and the other to multiple dynamic factors. These ratings were highly correlated with similarity judgments from a previous study. A second experiment assessed if the dynamic factors indirectly contributed to streaming by influencing perceptual attack time. The perceptual attack time of each tone was measured, and new sequences were recorded with the perceptual attacks isochronous. The results were highly correlated with the first experiment, and the effect of the dynamic factors was not diminished. These experiments indicate that streaming and similarity judgments are based on the same acoustic attributes. Implications for the segregation of melodic lines in music will be discussed.

10:45

2PP8. Perceptual segregation of tones embedded in modulated noise maskers. Punita G. Singh and Albert S. Bregman (Dept. of Psychol., McGill Univ., Montreal, Quebec H3A 1B1, Canada)

Maskers comprising four noise bands derived from same or different noise sources, and multiplied by same or different modulators, were compared for their efficacy in enabling perceptual segregation of a tone embedded in one of the bands. A two-interval paradigm was used, with a 1-kHz target tone followed by two noise bursts. The target tone was either absent, or present in the first, second, or both noise intervals at a fixed intensity level. The noise bands were centered at 0.5, 1, 1.5, and 2 kHz. A five-category labeling task was used. Labels "1," "2," and "3"

enabled subjects to report if the target tone was audible as a distinct perceptual entity in the first, second, or both noise intervals, respectively. Label "4" allowed reports of timbre change for cases where the cue tone was not perceived as a distinct entity, but the noises seemed different. Label "5" implied that the noise bursts seemed identical with no tones or timbre change perceived. Data for ten normally hearing listeners indicate that maskers with coherence of modulator envelopes across the four bands enabled better perceptual segregation of the tone, in accordance with predictions based on results from CMR experiments. The envelope-coherence cue overshadowed other variables such as masker bandwidth or sameness or difference of noise source. However, an interesting perceptual consequence was noted for the narrowest bandwidth (12.5-Hz) maskers. These tonal noises were more likely to be labeled as containing the target, despite its physical absence in the stimuli. While successful perceptual segregation of the target tone can be attributed to factors related to auditory grouping such as similarity of envelopes within the masker, it seems that other factors pertinent to grouping, such as similarity of features of the target and masker also play a role. [Work supported by NSERC.]

11:00

2PP9. The subtractive nature of illusory continuity: Reciprocal changes in alternating sounds. Richard M. Warren, James A. Bashford, Jr., and Eric W. Healy (Dept. of Psychol., Univ. of Wisconsin—Milwaukee, Milwaukee, WI 53201)

When portions of one sound are replaced by a louder sound, the missing segments may be restored if the louder inducing sound is a potential masker. This phenomenon has been examined extensively using pulsation threshold, auditory induction, and phonemic restoration paradigms to measure the conditions required for perceptual synthesis of the fainter sound. These studies have generally neglected to examine changes produced in the inducing sound. Decreases have been measured in apparent level of the inducer using inducer/inducee pairs consisting

of tone/tone, noise/noise, and noise/speech. Interestingly, changes in the inducer occurred even when the inducee was above the pulsation threshold. Under some conditions loudness reduction was coupled with a marked alteration in timbre: Especially profound changes in the quality of the inducer occurred when the alternating sounds differed only in intensity (e.g., two levels of a 1000-Hz sinusoidal tone)—apparently, a residue with anomalous neural representation remains following subtraction of components corresponding to the inducee. These observations involving the inducer require modification of current theories concerning the apparent continuity and restoration of absent sounds. [Work supported by NIH Grant No. DC00208.]

11:15

2PP10. A perceptual representation of sound for source separation. Daniel P. W. Ellis and Barry L. Vercoe (MIT Media Lab Music & Cognition Group, E15-491, Cambridge, MA 02139)

Building machines that emulate the kinds of acoustic information processing that human beings perform effortlessly has proved unexpectedly difficult. The irresistible conclusion is that the human auditory system is extremely sophisticated in its adaptation to real-world sounds and uses an impressive array of features as cues to organization and interpretation. As more of these cues become known through psychoacoustical experiment, it becomes feasible to program computers functionally to mimic human perception of sound. Since the processing is so integrated and the roles of different features only partly understood, the best approach to simulating real listeners (including susceptibility to illusions) is to build as direct an analog of the actual processing chain as can be devised. Concentrating on the task of distinguishing and separating individual superimposed sonic sources, such as a singer and accompaniment, a perceptually sufficient invertible representation has been built that analyzes sound down to a relatively small number of "atoms." Psychoacoustic rules of stream formation can then be applied to these atoms to simulate many aspects of human source separation. Examples will be shown and played.

TUESDAY MORNING, 12 MAY 1992

SALON H, 8:00 A.M. TO 12:35 P.M.

Session 2SA

Structural Acoustics and Vibration: Memorial Session in Honor of Eugen Skudrzyk

Sabih I. Hayek, Chair

Applied Research Laboratory and Graduate Program in Acoustics, Pennsylvania State University, State College, Pennsylvania 16804

Chair's Introduction—8:00

Invited Papers

8:05

2SA1. Broadband sound absorbers based on the mean-value theory. Alan D. Stuart (Appl. Res. Lab. and Graduate Prog. in Acoust., P.O. Box 30, Penn State Univ., State College, PA 16804)

The mean-value theory was developed by Professor Skudrzyk as a method for predicting and understanding the dynamic response of complex vibratory structures, like plates and shells [E. J. Skudrzyk, *J. Acoust. Soc. Am.* 67, 1105-1135 (1980)]. In this work he refers to the application of the theory to "situations that call for prescribed frequency response with a small number of resonators, e.g., matching absorbers to the wave resistance of the medium" [E. J. Skudrzyk, *Acustica* 64, 123-147 (1987)]. This paper will review the salient features of the mean-value theory and elaborate on how the author and Professor Skudrzyk applied the theory to the design of broadband sound absorbers.

8:30

2SA2. The beginnings of near-field acoustical holography under Eugen Skudrzyk. Earl G. Williams (Naval Res. Lab., Code 5137, Washington, DC 20375-5000)

Near-field acoustical holography (NAH) had its roots in Eugen Skudrzyk's laboratory in the Davey Building at Penn State. He inspired construction of the first near-field array of microphones, with 256 elements in a large square array, placed in his semianechoic chamber in the Davey Laboratory. This chamber had been used by several previous students, including myself, to study plate vibration. In another room, Graham and Watson, two former Ph.D.'s of Eugen Skudrzyk, had completed work years before with a scanning microphone and had developed a technique called acoustical holography, using a laser and photographic film to produce reconstructions of the sound field. With this new array of microphones, especially with a computer interface, it was possible to digitally process and reconstruct the recorded hologram data eliminating the need for lasers and photographic film and improving tremendously the quality of received data. In this talk the work in NAH at the Penn State University, and the inspirational role which Eugen Skudrzyk played, will be discussed.

8:55

2SA3. Aeroacoustics of transitional and separated boundary layers. Gerald C. Lauchle (Appl. Res. Lab. and Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Experimental and theoretical research related to the acoustic emissions from nonsteady laminar and turbulent boundary-layer flows is described. Transition of a laminar boundary layer into a turbulent one is characterized by the formation and growth of turbulent spots. The spots are interspersed with laminar regimes (thus causing the laminar boundary layer to be nonsteady) and they create local wall pressure fields and acoustic radiation that are notably different from those generated by a full turbulent layer. The current research in this arena is concerned with refining the theoretical bases for predicting the acoustic radiation and on the performance of experiments to help verify these models. Other research deals with the radiation and structural response due to the separated turbulent flow over forward and rearward facing steps on flat panels. This effort is experimental with some new analytical techniques being developed for the signal processing of pressure and intensity probe signals that permit identification of coherence areas and source strengths for the reattaching flow. The research projects discussed here are forming the dissertations of R. C. Marboe and W. A. Kargus, Ph.D. candidates in the Graduate Program in Acoustics at Penn State. [Work supported by ONR and Ford Motor Co.]

9:20

2SA4. Underwater polyvinylidene fluoride (PVDF) acoustic sensors. W. Jack Hughes (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

Polyvinylidene fluoride (PVDF) is increasingly being used as a piezoelectric sensor in the construction of underwater hydrophones. Its desirable properties of high sensitivity, small thickness, mechanical flexibility, and variable electrode geometry, made PVDF an intriguing sensor to develop. One of the first applications of PVDF at ARL/PSU was in the measurement of the flow noise over the surface of an underwater vehicle. These data were compared with flow noise data that had been gathered by Professor Skudrzyk in the late 1960's. Dr. Skudrzyk's most recent interest in flow noise led to the conduction of field tests of three differently shaped PVDF sensors. These data will be presented. Also to be discussed are several of the numerous applications that PVDF has been put to at ARL/PSU: underwater intensity probes, hydrofoil pressure fluctuation monitoring, annular ring beam forming, large area/high-frequency sensors, and shaped-beam element sensors. Some of the advantages, disadvantages, and construction problems with PVDF sensors will also be addressed.

9:45

2SA5. Subsea vehicle pump pulsation measurements. Murray M. Simon (Alliant Techsystems, Inc., Mukilteo, WA 98275)

The use of seawater ballast pumps for small undersea vehicles has increased demands for noise reduction in such subsystems. With severe constraints on volume, weight, and efficiency and the need to operate at high head pressure, the usual noise reduction techniques used on large submarines are not feasible. There are stringent requirements on a measuring system to evaluate such subsystems. To prevent biased results due to the differing spectral content of various subsystems, nonreactive acoustic termination is required. Also, care must be taken to ensure that the hydraulic load providing the required back pressure does not contaminate the results with its own noise. This paper describes an easily assembled system meeting the requirements for testing in a laboratory environment. Data were taken on two styles of water pumps to evaluate the system. Two in-line acoustic filters (mufflers) of the commercially available "three pipe" configuration were tested with one of the pumps and compared to analytical results using a transmission matrix model. The results indicate significant insertion loss over a broad frequency range. [Work supported by Alliant Techsystems internal IR and D funding.]

2SA6. Quantitative schlieren analysis of acoustic interaction with submerged plates. Roger T. Richards (Naval Undersea Warfare Ctr., New London, CT 06320-5594)

Transmission and radiation of sound from submerged plates were investigated by means of a quantitative schlieren method that produced accurate visual representations of sound fields. The method was shown to be significantly faster than the usual point-by-point scans of acoustic fields with no loss of accuracy. The low-frequency resolution of the classical schlieren system was improved by nearly two orders of magnitude; sound waves with frequencies as low as 27 kHz were easily visualized. These improvements were demonstrated with a thorough investigation of sound transmission through submerged plates over a frequency range from well below to over one hundred times the classical coincidence frequency. Beam shifting caused by Lamb waves and a previously unknown cancellation with increasing frequency of modal pairs of Lamb waves were both observed.

Contributed Papers

10:35

2SA7. Response of a spherical shell to incident sound in water at high frequency. Robert Hickling and James F. Ball (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, Coliseum Dr., University, MS 38677)

Continuum-theory computations show that, for high-frequency continuous-wave excitation, the elastic waves in the shell structure separate into surface waves at the inner and outer surfaces of the shell, particularly at the back of the shell away from the incident sound. The computations also show how the elastic waves depart from consistency with thin-shell theory, at low to intermediate frequencies, as frequency increases. At high frequencies, the principal reaction occurs at the front of the shell closest to the sound source. Further exploration of high-frequency behavior is needed, particularly as it relates to ray theory. It is necessary also to relate the elastic waves in the structure to the scattered sound field, especially through use of resonance scattering theory.

10:50

2SA8. Variational analysis of thin beam excitation using asymmetric piezoelectric actuators. Guy Plantier and Jean Nicolas (G.A.U.S., Dép. de génie mécanique, Univ. de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

This paper presents a dynamic model as well as an experimental study of the response of a beam to excitation by a single piezoelectric ceramic glued to the structure. Such an actuator, when excited, produces both flexural and extensional motion in the structure. Based on an analytical approach using a variational method, the model takes into account the dynamic coupling between the structure and the actuator, includes the free stress conditions at the piezoelectric actuator boundaries and allows one to predict with accuracy the beam displacement and thus the extensional strain on the surface of the structure for any beam boundary condition. Although essentially devoted to the study of asymmetric piezoelectric actuators, the accurate model presented here allows one to discuss the simplifying hypothesis usually made in past studies on symmetric actuators.

11:05

2SA9. Performance capabilities of the adaptive filtered-x algorithm for active control of broadband excitation. Scott D. Sommerfeldt and Peter J. Nashif (Appl. Res. Lab. and Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Applications of the adaptive filtered-x algorithm to active noise and vibration control problems involving broadband excitation of the system have been reported previously in the literature. For these different applications, varying degrees of control effectiveness have been reported,

sometimes for the same class of active control problem. This paper will address the issue of using the filtered-x algorithm for broadband control applications to establish the performance that can be expected based upon the parameters associated with the system to be controlled. In particular, issues such as causality of the control system, the characteristics of the broadband excitation, and the amount of damping in the physical system will be addressed to determine how such parameters can be expected to influence the control achieved.

11:20

2SA10. Experimental measurement of structural intensity in the frequency-wave-number domain. S. Padiyar and J. M. Cuschieri (Ctr. for Acoust. and Vib. Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

Typical methods of structural intensity measurement involve the use of either contacting transducers or single and dual channel laser vibrometers. Because of the generally limited number of measurement locations, approximations that can limit the accuracy of the measured structural intensity data are implemented. The accuracy of the data can be compromised especially in the case of finite structures. Furthermore, if waves with different wave-number components exist, the measurement schemes used are typically tailored toward only one wave type. Using a multiple channel laser vibrometer, with more than two channels, eliminates the need for the approximations in the measurements. Additionally, by scanning the surface of the structure, the structural intensity components can be decomposed by using frequency-wave-number transforms. The result for the structural intensity in the frequency-wave-number domain will discriminate the intensity component into those associated with in-plane and out-of-plane waves. The frequency and wave-number transforms are obtained using digital signal processing techniques. One disadvantage with this approach is that inherent in discrete signal processing. In this paper, issues associated with the range of frequency and wave number(s), spatial window size, selected window type, and method of processing are discussed. A data processing scheme for frequency-wave-number measurements of structural intensity on finite structures is defined. Results showing the application of this technique using the multiple channel laser vibrometer will also be presented. [Work sponsored by ONR.]

11:35

2SA11. Passive adaptive control of plate vibration and acoustic radiation. Deborah A. Gruenhagen, Courtney B. Burroughs, and Scott D. Sommerfeldt (Graduate Prog. in Acoust., Penn State Univ., State College, PA 16804)

The frequencies of resonance of the lower-order modes in a plate are sensitive to boundary conditions. By adaptively controlling the clamp-

ing pressures on the edges of a vibrating plate, the frequencies of resonance can be continuously detuned from the frequency of a pure-tone source. Results from laboratory demonstrations of the passive adaptive control of the vibration and acoustic radiation from a plate excited by pure-tone sources with varying source frequencies are presented and discussed.

11:50

2SA12. Time optimal active damping of a cantilever beam. Yichiang Syang and Ilene J. Busch-Vishniac (Dept. Mech. Eng., Univ. of Texas, Austin, TX 78712)

Active damping control of spatially distributed systems has typically been accomplished using a number of discrete sensors and actuators. However, use of discrete elements introduces the problem of actuator/observer spillover that can be quite a problem for large lightly damped structures, such as might be used in space applications. While some researchers have studied distributed sensing and actuation, their application to active damping of structures has been limited primarily to a Liapunov energy minimization approach, in which the control strategy is derived by making the power always flow out of the vibrating structure. In this work modal sensors and actuators such as described by Lee and Moon [J. Acoust. Soc. Am. 85, 2432–2439 (1989)] are used in the design of an active damping system for a cantilever beam. The modal sensors and actuators control the lowest modes of the system and produce no spillover. A time optimal control strategy, which enables one to minimize the time it takes to make the vibration cease, is used. This control method, while more difficult to implement than Liapunov control algorithms, directly addresses the desired objective of damping the vibration as fast as possible.

12:05

2SA13. Bending waves due to a moving harmonic force. Mauro Pierucci (Dept of Aerosp. and Eng. Mech., San Diego State Univ., San Diego, CA 92182)

In many structurally induced and flow-induced vibration problems, the harmonic forcing function is not stationary but moves with a velocity V_0 . The effect of the forcing function velocity V_0 upon the free vibrational wave-number characteristics of a membrane and a plate is

analyzed. The Mach number M is defined to be the ratio of the velocity V_0 to the wave speed of the bending waves. For the membrane, the effect of the Mach number is to increase the wave number (shorter wavelength) ahead of the forcing function and to decrease the wave number (longer wavelength) behind it. At supersonic speeds no disturbances travel ahead of the forcing function and both wave numbers lead to trailing waves. These results are equivalent to the classical Doppler-shifted results. The results of the plate are more complex. The right and left traveling waves retain their basic properties with the magnitude of the wave number changing monotonically as a function of the Mach number M . The near-field decaying disturbances also retain their basic properties, but immediately obtain components that induce the decaying disturbances to become left traveling waves with decaying components. At Mach numbers greater than 2, these disturbances become pure waves trailing without any decaying factor. The importance of each of these components as a function of the Mach number is discussed.

12:20

2SA14. An automated focusing and scanning system for measuring fast waves on elastic structures by laser Doppler interferometry. H.-G. Kil, Jacek Jarzynski, and Yves H. Berthelot (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

A laser Doppler vibrometer has been designed, built, and tested for specific applications to structural acoustics. It is capable of measuring small in-plane displacements associated with the propagation of fast waves (shear or longitudinal) on the surface of a structure. The probe head is compact, rugged, and can operate underwater. The system has been automated so that extensive data can be collected over the structure and analyzed in the K -space as is done in acoustical holography. The motion of the probe head is controlled by two stepping motors. The first one controls the focusing action (laser spot size on the structure) and the second one controls the position of the whole probe over the structure. The laser Doppler signal measured by the probe head is detected by a photodiode, demodulated by a phase-lock loop, digitized, and sent to a computer that acquires the data and controls the stepping motors. Automated focusing action is achieved by an iterative process. Automated scanning over a given structure is also achieved digitally by computer control of the second stepping motor. Results obtained with the system collecting data over a vibrating surface will be presented. [Work supported by ONR.]

Session 2SP

Speech Communication: Nonphonetic Influences on Speech Organization and Intelligibility

Sharon Y. Manuel, Chair

Wayne State University, CDS, 563 Manoogian Hall, Detroit, Michigan 48202

Chair's Introduction—8:10

Contributed Papers

8:15

2SP1. Phonological representation affects phonetic perception. Allard Jongman, Joan Sereno (Dept. of Modern Languages and Linguistics, Cornell Univ., Ithaca, NY 14853), Aditi Lahiri, and Marianne Raaijmakers (Max Planck Inst. for Psycholinguistics, Wundtlaan 1, 6525 XD Nijmegen, The Netherlands)

This study examines to what extent phonological representations affect word identification in Dutch. Dutch has an underlying contrast in voicing that is neutralized word-finally. Also, vowels are lengthened approximately 20 ms when preceding medial voiced consonants. Vowel length can therefore be a cue to voicing. The present study investigates whether the vowel length cue influences listeners when hearing stimuli with ambiguous vowel duration in an identical, neutralized, consonantal context but where the underlying representation of the consonant differs in voicing. A vowel length continuum ([at] to [a:t]) was made by shortening a long vowel in 12 steps. To this continuum, initial consonants were added to create four phonetic endpoints with opposite underlying voicing patterns: /zat/, /zaad/, /stad/, /staat/. Results of a vowel categorization task showed that, using an identical vowel length continuum, the crossover boundary in the *zat-zaad* continuum occurs at a significantly shorter vowel length than that in the *stad-staat* continuum. These results provide evidence that listeners use the underlying phonological representation in the perception and identification of words.

8:30

2SP2. Effect of levels of stimulus uncertainty and consonantal context on formant frequency discrimination. Diane Kewley-Port (Dept. of Speech and Hear. Sci., Indiana Univ., Bloomington, IN 47405)

Discrimination thresholds for the second formant of vowels in consonantal context were significantly larger than for vowels in isolation in a report by Mermelstein [J. Acoust. Soc. Am. 63, 572 (1978)]. These results were not upheld in a recent report to this society [Kewley-Port and Watson, J. Acoust. Soc. Am. 89, 1996 (A) (1991)] for testing under minimal stimulus uncertainty procedures. Moreover, Mermelstein's thresholds were a factor of 5 larger than those reported by Kewley-Port and Watson. The present experiment examines the effects of testing under higher levels of stimulus uncertainty in conditions more similar to those of Mermelstein. For well-trained subjects, most thresholds for the eight CVC stimuli tested did not change when levels changed from minimal to medium uncertainty. Further studies, using testing conditions similar to those of Mermelstein, employed untrained subjects. Results indicated that some subjects rapidly improved their ability to discriminate changes in formant frequency during just 1 h of testing. The differential effects of consonantal context, training, and

levels of stimulus uncertainty will be discussed. [Research supported by NIH and AFOSR.]

8:45

2SP3. Perceptual space and learning in speech perception. Sandra J. Guzman and Howard C. Nusbaum (Dept. of Psychol., Univ. of Chicago, 5848 S. University Ave., Chicago, IL 60637)

When listeners are given training on synthetic speech, intelligibility improves and listeners become more efficient in using cognitive resources for the perception of speech [L. Lee and H. C. Nusbaum, J. Acoust. Soc. Am. Suppl. 1 85, S125 (1989)]. The present study investigated one possible account of the connection between changes in intelligibility and changes in efficiency in using cognitive capacity: Learning may shift cognitive resources away from the analysis of acoustic properties of synthetic speech that are not informative about phonetic structure to more informative acoustic properties. If learning changes the distribution of attention to the acoustic signal, a change should occur in the structure of the perceptual space used in recognizing synthetic speech. Multidimensional scaling was performed on consonant confusions before and after training in order to determine if and how the structure of perceptual space is affected by learning. The results are consistent with the hypothesis that perceptual learning of synthetic speech changes the way listeners focus attention on the acoustic properties of the speech signal. [Work supported by NIDCD.]

9:00

2SP4. On the weakness of using strong syllables as word boundary markers. Paul N. Yerkey and James R. Sawusch (SUNY at Buffalo, Amherst, NY 14260)

Previous research has shown that a strong-weak syllable distinction may play an important role in word segmentation. Cutler and Norris [JEP:HPP 14, 113-121 (1988)] asked subjects to identify words at the beginnings of two syllable nonwords. Subjects were faster to identify a word when the second syllable was weak than when it was strong. The present study included lax vowels in addition to the tense and neutral vowels previously used to form the second syllables. The lax vowel produces a strong syllable of short duration; something not previously present. By comparing word identification for items in which the second syllable is strong with either a tense or lax vowel to the weak syllables, the relative roles of strong versus weak syllables and vowel duration can be explored. To the extent that tense and lax vowel syllables produce equivalent effects, strong syllables act as a cue to word boundaries in English. [Work supported by NIDCD Grant No. DC00219 to SUNY at Buffalo.]

2SP5. Novel findings concerning intelligibility of bandpass speech. Keri R. Riener, Richard M. Warren, and James A. Bashford, Jr. (Dept. of Psychol., Univ. of Wisconsin—Milwaukee, Milwaukee, WI 53201)

Very high intelligibility was found for a number of narrow-band filtering conditions. Four hundred listeners (20 groups of 20 subjects) were presented with bandpass filtered CTD sentences ("everyday speech") and monosyllabic words. Separate groups received center frequencies of 370, 530, 750, 1100, 1500, 2100, 3000, 4200, or 6000 Hz at 70 dBA SPL. In experiment 1, intelligibility of single 1/3-octave bands with steep filter slopes (96 dB/oct) averaged more than 95% for sentences centered at 1100, 1500, and 2100 Hz, and more than 50% for monosyllabic words centered at 1500 and 2100 Hz. Experiment 2 used the same center frequencies with extremely narrow bands (slopes of 115 dB/oct intersecting at the center frequency)—intelligibility remained relatively high for most bands, with greatest intelligibility at 1500 Hz (77% for sentences, 18% for words). In experiment 3, 1/3-octave bands (96 dB/oct) centered at 370 and 6000 Hz were presented simultaneously either diotically or dichotically (when presented separately in experiment 1, intelligibility of these bands did not differ significantly, and averaged 23% for sentences and 3% for monosyllabic words). When the bands were combined, diotic and dichotic presentations were equivalent. Intelligibility rose to an average of 77% for sentences and 34% for words. [Work supported by NIH Grant No. DC00208.]

9:30

2SP6. Acoustic and articulatory predictors of p-center perception. Kenneth de Jong (Phonetics Lab. Dept. of Linguistics, UCLA, Los Angeles, CA 90024-1543)

To align syllables rhythmically, listeners do not use their acoustic onsets but rather some internal point, called the p center, or perceptual moment of occurrence of the syllable. Fowler [J. Exp. Psychol. Gen. 112, 386–412] claimed that p centers are associated with the timing of

the underlying speech gesture for the vowel in the syllable. Marcus [Percept. Psychol. 30, 247–256], using tokens whose variation was produced by digitally editing a target word, found that the acoustic factors affecting the location of the p center were distributed throughout the syllable, suggesting that p-center locations are a function of the entire content of the syllable. Presently, a replication of Marcus's experiments is being conducted, which uses naturally varied stimuli. Variation was introduced by eliciting tokens in different discourse settings. To determine the relationship of p-center location to articulatory events, articulatory recordings of the tokens were made with the Wisconsin x-ray microbeam system. This paper will present the results of correlations between p-center location responses and various acoustic and articulatory predictors of p-center location. [Work supported by the NIH and NSF.]

9:45

2SP7. Effects of base complexity in musical duplex perception. Michael D. Hall and Richard E. Pastore (Psychoacoust. Lab. and the Cognitive and Psycholinguistic Sci. Ctr., Dept. of Psychol., SUNY at Binghamton, Binghamton, NY 13902-6000)

Duplex perception (DP) occurs when one stimulus or stimulus component contributes simultaneously to two distinct percepts. Most DP research has been used to loosely address the postulation of a phonetic module. The present musical DP experiments focus instead on the utility of DP in evaluating perceptual organization. The present investigation evaluated one variable which, according to Gestalt notions, could be critical to DP, base complexity (the number of frequency components in the base). In a modified AX task, two experiments used chord stimuli, dichotically presenting a fixed base and variable (major/minor chord) distinguishing tone on each trial. Fusion of contralateral distinguishing tones was frequent, but contrary to modularity conjectures, little evidence was found for triplex perception. Other unexpected results will be presented. Findings are discussed as they apply to (a) speech stimuli (stimulus dominance) and (b) attentional constructs (as an example of feature integration). [Work supported by NSF.]

10:00–10:15

Break

10:15

2SP8. Syllabic perception: Effects of absolute and relative temporal cues. Charles A. Harsin and Robert J. Porter, Jr. (Dept. of Psychol., Univ. of New Orleans, New Orleans, LA 70148 and Kresge Hear. Res. Lab. of the South, Dept. of Otorhinolaryngology, Louisiana State Univ. Med. Ctr., New Orleans, LA 70122)

To investigate how syllabic differences might be conveyed by the manifestations of articulator gestures in the absolute and relative timing of acoustic cues, speech signals corresponding to the V + C + V sequence /ipi/ were created. The durations of various acoustic cues were independently manipulated, and subjects were asked to judge whether particular signals had the structure /i#pi/ or /ip#i/. Subjects were also asked to discriminate among the same signals. A clear, "categorical" relationship was found between bursts' relative temporal positions in the intervocalic interval and syllabic categorization. On the other hand, discrimination measures revealed sensitivity to small absolute differences in segment durations regardless of judged syllable type. Thus a comparison of syllable judgment and discrimination results suggests

that both absolute and relative temporal cues are available for perceptual decisions but that relative cues are selected for syllable judgments, perhaps because they map more directly onto the relative timing of articulatory gestures. [This research was supported in part by University of New Orleans Chancellor's Fellowship.]

10:30

2SP9. Further results on the duration of infrequent and frequent words. D. H. Whalen (Haskins Labs., 270 Crown St., New Haven, CT 06511)

In a study reported at the last meeting of the Society, subjects produced longer durations when reading lists of infrequent words than ones with frequent words. This was in spite of the fact that the words contained exactly the same phonemes by virtue of being homophones (e.g., "right" and "rite"). As in that study, subjects in the present experiment read rehearsed lists as quickly as they could, but this time, the two frequency types occurred in each list. Preliminary results show that, as

before, the infrequent items were longer in duration, but the magnitude was reduced compared to the earlier results. Since these were the same subjects in the same session, direct comparisons are possible. Several explanations for the difference will be explored. First, the frequency category of the following item might affect duration as well. This effect, found in lexical access studies, may play a role here even though the lists were rehearsed. Second, subjects may simply have spoken more quickly in the later part of the session; the previous results had suggested that the frequency effect did not hold for the fastest talkers. Finally, it may simply be that by the time the subjects had uttered these words ten times, they were all temporarily higher in frequency. Whatever the explanation for the difference, the continued presence of an effect suggests, as before, that a link to the lexicon is maintained throughout production. [Work supported by NIH Grant No. DC-00825.]

10:45

2SP10. Syllable internal timing: Effects of speaking rate, sentence position, and focal stress. Dawn M. Behne and Lynne C. Nygaard (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

The duration of a vowel can be affected by speaking rate, sentence position, and focal stress. Having confirmed the independent effects of these factors on vowel duration, the goal of this study was twofold: (1) to describe their effects on the timing of consonants neighboring the affected vowel, and (2) to characterize the internal timing of the syllable when these factors are combined in the same sentence. Conversations were developed in which target CVCs occurred in initial or final sentence position and were either focused or nonfocused by the discourse. Twelve subjects produced each conversation at three speaking rates. The results indicate that (1) like vowels, consonant durations are affected by speaking rate, sentence position, and focal stress, and (2) when converging on a syllable, the investigated factors influence the syllable's internal timing independently. These findings suggest that speaking rate, sentence position, and focal stress each globally affect the internal timing of a syllable, with the effects of each factor being superimposed on one another within flexible upper and lower limits.

11:00

2SP11. The effects of speaking rate and amplitude variability on perceptual identification. Mitchell S. Sommers, Lynne C. Nygaard, and David B. Pisoni (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

Considerable evidence now exists [J. L. Miller, *Phonetica* 38, 159–180 (1981)] that the temporal properties of speech signals convey phonetic information in a rate-dependent manner. Although most recent models of speech perception have incorporated a stage at which listeners perceptually compensate for variations in articulation rate, as well as other linguistically important sources of variability, they have universally failed to specify whether there are perceptual costs associated with this “normalization” process. The present study was designed to determine whether open-set word identification is influenced by two sources of naturally occurring variability in the speech signal—differences in rate of articulation and overall stimulus amplitude. These two types of variability were chosen because the former is known to be important for phonetic judgments while the latter is generally considered phonetically irrelevant. The results demonstrated that identification of monosyllabic words was significantly poorer when the items were presented in lists containing three, as opposed to one, speaking rates. In contrast, identification accuracy was not significantly affected by the introduction of variability in overall amplitude. These findings extend the results of previous investigations [J. W. Mullenix and D. B. Pisoni, *J. Acoust. Soc. Am.* 85, 365–378 (1989)] that demonstrated detrimental effects of talker variability for word recognition and further suggest that only variations along phonetically relevant dimensions are likely to result in poorer identification performance. [Work supported by NIH.]

11:15

2SP12. Effects of rate and talker variability on the recall of spoken words. Lynne C. Nygaard, Mitchell S. Sommers, and David B. Pisoni (Speech Res. Lab., Dept. of Psychol., Indiana Univ., Bloomington, IN 47405)

Changes in speaking rate and talker characteristics have profound effects on the acoustic structure of the speech signal. Current models of speech perception hypothesize that listeners must compensate for changes in rate and talker with a normalization process in which phonetic segments are evaluated relative to the prevailing rate of articulation and relative to specific talker characteristics. The present study was conducted to investigate how listeners adjust or normalize for changes in rate of speech and talker characteristics. A serial recall task was used to evaluate the consequences of rate and talker variability on the early encoding and rehearsal of the spoken words. Listeners were asked to recall words from single versus multiple rate lists; from single versus multiple talker lists; and finally, from lists with words produced by multiple talkers at multiple speaking rates. The results showed that words from lists produced at a single articulation rate were recalled more accurately in early serial positions than words from lists produced at multiple articulation rates; words from lists produced by a single talker were recalled more accurately in early serial positions than words from lists produced by multiple talkers; and finally, the combination of variability in talker and rate did not impair recall relative to lists with variation in rate or talker alone. These findings replicate and extend the results of previous research on talker variability [C. S. Martin *et al.*, *J. Exp. Psych.: Learn. Mem. Cog.* 15, 676–684 (1989); S. D. Goldinger *et al.*, *J. Exp. Psych.: Learn. Mem. Cog.* 17, 152–162 (1991)] suggesting that variations in speaking rate and in talker characteristics incur a processing cost that affects the initial encoding and subsequent rehearsal of spoken words. [Research supported by NIH.]

11:30

2SP13. Prominence caused by rising and falling pitch movements with different positions in the syllable. Dik J. Hermes and H. H. Rump (Inst. for Perception Res./IPO, P.O. Box 513, NL 5600 MB Eindhoven, The Netherlands)

The object of this study was to investigate whether subjects are able to compare the prominence caused by different types of accent-lending pitch movements, and, if so, whether some pitch movements lend more prominence to a syllable than others. These experiments were carried out with the utterance/mamáma/, with the second syllable accented by either a rise, a fall, or a rise–fall. Subjects adjusted the variable excursion size of a comparison stimulus to the fixed excursion size of a test stimulus in such a way that the accented syllable in test and comparison stimuli had equal prominence. The rise–fall was only presented in standard position, the fall and the rise were tested for five different positions in the syllable. It is concluded that subjects are well able to equate the prominence of syllables accented by various types of pitch movement, viz., a rise–fall in standard position, a rise starting before the vowel onset, and a fall whatever its position in the syllable. Moreover, when lending equal prominence, the early starting rise and the rise–fall have equal excursion sizes. The fall, however, appears to lend more prominence to a syllable than the rise or the rise–fall of equal excursion size, independent of its position in the syllable. This difference increased with increasing declination of the pitch contour.

11:45

2SP14. Individual variability in the perception of natural and synthetic speech. Valerie L. Hazan and Bo Shi (Dept. of Phonetics and Linguistics, Univ. College London, 4 Stephenson Way, London NW1 2HE, UK)

The extent of individual variability in the perception of natural and synthetic speech was examined in a group of 60 listeners, homogeneous

in terms of age, language, background, hearing threshold, and exposure to synthetic speech. Listeners were tested on three types of speech material, representing different levels of contextual information: nonsense syllables (VCV), semantically unpredictable sentences (SUS), and SPIN sentences, in which test words are presented in high- and low-probability sentence contexts. All tests were presented in synthetic speech and natural speech-in-noise conditions. A large amount of variability in scores was found for all tests. Cluster analyses showed the presence of three main listener groups, differing mainly in their perfor-

mance on low-redundancy sentence material. There was a strong correlation between the SUS and SPIN sentence test results but performance on synthetic speech was not strongly correlated with performance on natural speech in noise. Scores obtained for word and sentence tests were compared to the listeners' performance on reduced-cue identification tests for place and voicing contrasts in order to ascertain whether there was a link between a listener's use of acoustic cues and need for contextual redundancy. [Work supported by SERC.]

TUESDAY MORNING, 12 MAY 1992

SALON D, 7:55 A.M. TO 12:00 NOON

Session 2UW

Underwater Acoustics: Scattering and Reverberation

Michael A. Schoenberg, Chair

Schlumberger Cambridge Research, Madingley Road, Cambridge CB3 0EL, England

Chair's Introduction—7:55

Contributed Papers

8:00

2UW1. The small slope approximation for acoustic scattering at a fluid-solid interface. Shira L. Broschat and Taiqian Yang (School of Elec. Eng. & Comput. Sci., Washington State Univ., Pullman, WA 99164-2752)

Acoustic scattering from the ocean bottom is of much interest to the underwater acoustics community. Recent results for the small slope approximation for one- and two-dimensional randomly rough pressure-release surfaces have been promising. In this paper, work on the small slope approximation is extended to acoustic scattering from a fluid-solid boundary for one-dimensional, randomly rough surfaces with a Gaussian roughness spectrum. Expressions are derived for the scattering strength using both the first- and second-order small slope approximations. Numerical results for the first-order approximation using parameters characteristic of acoustic scattering from a water-granite interface are presented. These results compare well with those obtained by Berman and Dacol [J. Acoust. Soc. Am. **84**, 292-302 (1988)] using perturbation theory when it is expected to give accurate results. They contain the same interesting structural features at the critical angles for the transmitted compressional and shear waves. [Work supported by ONR.]

8:15

2UW2. The local parabolic approximation for rough surface scattering. Kristinka Ivanova and Shira L. Broschat (School of Elec. Eng. and Comput. Sci., Washington State Univ., Pullman, WA 99164-2752)

A method, based on a technique introduced by A. V. Belobrov and I. M. Fuks [Radiophys. Electron. (Izv. VUZ Radiofiz.) **29** (12), 1083-1089 (1986)], is studied for the case of scalar wave scattering from one-dimensional randomly rough surfaces with a Gaussian roughness spectrum. Using a local parabolic approximation of the rough surface, the surface source density is obtained. The small parameter in such an

approximation is the inverse of the radius of the surface curvature multiplied by the wave number and the cube of the cosine of the local angle of incidence. The bistatic scattering cross sections per unit length are derived for the Dirichlet and Neumann boundary conditions. The first term for each is that of the Kirchhoff approximation; successive terms account for diffraction effects. Numerical results for both soft and hard surfaces are obtained and compared with other available numerical results. [Work supported by NSF and ONR.]

8:30

2UW3. Renormalization group analysis of rough surface scattering. G. J. Orris¹ and R. F. Dashen (Univ. of California, San Diego, Phys. Dept. 0350, 9500 Gilman Dr., La Jolla, CA 92037-0350)

An acoustic field of wavelength much larger than the largest scale of a scattering body is considered. A renormalization group equation has been developed to find an easier way of calculating the ensemble averaged scattered fields, with the renormalized parameter being impedance constant introduced to describe the behavior of the boundary under insonification. The results suggest that rough surface scattering is, in fact, an asymptotic theory, under the limit that the surface structure has no lower cutoff and the scatterer has a power-law surface spectrum. Also, a fundamental difference in the renormalization scheme between the two extreme boundary conditions (von Neumann and Dirichlet) is pointed out. ¹Currently at Naval Research Laboratory, Washington, DC 20375-5000.

8:45

2UW4. Operator expansion error estimation by higher-order terms. John Dubberley (NRL, Code 221, Stennis Space Center, MS 39529-5004)

The operator expansion method for quickly solving the Helmholtz

boundary value equation was outlined by Milder [J. Acoust. Soc. Am. **89**, 529–541 (1991)]. This method was verified by Kaczowski and Thorsos for impenetrable surface scattering where Kirchhoff and small perturbation theory hold [J. Acoust. Soc. Am. **90**, 2258 (A) (1991)]. In this presentation the third-order and higher terms of the symmetric operator expansion will be derived in order to place error bounds on the integral approximation for second-order and higher terms of the expansion. These error bounds can then be used to determine the regions of validity for the next lower-order expansion approximations. Some examples and comparisons to other scattering methods will be shown.

9:00

2UW5. An improved Kirchhoff approximation for scattering from penetrable media using the operator expansion. Kevin LePage (Dept. of Ocean Eng., MIT, Cambridge, MA 02139), Dajun Tang (Woods Hole Oceanographic Inst., Woods Hole, MA 02543), and Henrik Schmidt (MIT, Cambridge, MA 02139)

Recent work by Milder [J. Acoust. Soc. Am. **89**, 529–541 (1991)] has shown that increased efficiency in evaluating the Helmholtz integral equation for scattered fields may be obtained through a method known as an operator expansion (OE). More recently Kaczowski and Thorsos [J. Acoust. Soc. Am. **90**, 2258 (A) (1991)] have validated the accuracy of the OE for a broad variety of impenetrable 1-D rough surfaces. Here the OE formalism is used to obtain results for penetrable, even acoustic–elastic interfaces. Since in penetrable problems neither the value of the scattered field nor its derivative on the interface is known, it is necessary to seek some approximation. In this work a modified Kirchhoff approximation is proposed, where the scattered field on the interface is approximated by the incident field multiplied by the local reflection coefficient, but its derivative is approximated using the operator expansion. Two-dimensional scattered field realizations computed using this procedure are compared to exact fluid–fluid results obtained by a boundary integral approach and fluid–elastic results obtained by the boundary element method of Gerstoft and Schmidt [J. Acoust. Soc. Am. **89**, 1629–1642 (1991)]. Conclusions are drawn concerning the accuracy of the method and the feasibility of using it to estimate three-dimensional reverberation.

9:15

2UW6. A general theory of scattering by thin interface layers, plates, and shells. Gerry Wickham^{a)} (Ames Lab., Iowa State Univ., Ames, IA 50011)

In recent years there have been a number of attempts to model the effect of thin interface layers on the scattering and transmission of sound by bodies submerged in a continuous medium. Interface layers or coatings occur in numerous engineering acoustic applications. For example, when two solid bodies are bonded together, the adhesive or the imperfect fusion region will generally introduce an acoustic mismatch with both partners. Similarly, the use of austenitic cladding as a protection against corrosion creates an analogous problem, except that here the layer should be thought of as being fused to the surface of the material. In underwater and aeroacoustics the fluid–solid interactions with a shell or plate are often modeled using the Euler–Bernoulli thin plate theory. As far as the scattering problem here is concerned, the thin plate equation appears as an “effective boundary condition” relating the transverse displacement to the fluid pressure. Similarly, a thin interface layer be-

tween two solids is also modeled by some effective boundary condition. A typical approach is to assume that the layer is characterized by interfacial mass, spring, and damping constants. The latter are estimated by comparing the predictions of the model with the exact solutions to certain static canonical problems. The difficulty with this approach is that, generally, it is not self-consistent. In other words if the spring constant is calculated so that the predictions of the model agree well with certain exact solutions at “normal incidence,” say, then it is usually not the case that they give agreement under different loading conditions. Similarly, in the fluid–solid interaction problem, the Euler–Bernoulli theory gives a correct approximation to the dispersion relation for leaky antisymmetric Rayleigh–Lamb waves in a fluid-loaded thin plate but incorrectly predicts the transmission and reflection of a plane wave by such a plate in the fluid! In this paper these problems are addressed by developing a rational approximation scheme for an arbitrary thin layer immersed in a contrasting matrix.^{a)}The author is currently on leave from The Department of Mathematics, University of Manchester, Manchester M13 9PL, U.K.

9:30

2UW7. A new formulation for scattering of sound by cylinders with noncircular cross section. Daniel T. DiPerna (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

The problem of scattering of acoustic waves from a cylinder of noncircular cross section is treated using a plane wave expansion in Cartesian coordinates. By changing to a coordinate system in which “radial coordinate = constant” is the scatterer surface, series expansions for the incident and scattered field are obtained. The form of these expressions allows rigid or soft and in some cases penetrable (fluid) boundary conditions to be easily satisfied. The result is a system of equations that may be solved for the expansion coefficients of the scattered field. Numerical results for a 10:1 aspect ratio elliptic cylinder under rigid boundary conditions show excellent agreement with the exact solution. [Work supported by ONR.]

9:45

2UW8. Measurements of scattering by irregularly shaped bodies. Peter D. Thorne, Frank A. Boyle, Nicholas P. Chotiros, and Kendall R. Waters (Advanced Sonar Div., Appl. Res. Labs, Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

Scattering by spheres is a classical problem in underwater acoustics, and developments over the past two decades have established a detailed description of the scattering process in terms of the form function structure. More recently, interest has focused on sound scattering by spheroids and a number of studies have successfully applied the T -matrix formulation to solving the scattering problem. Currently the scattering of sound by irregular bodies is receiving increasing attention because of the application of acoustics in studying suspended sediment processes, the effect of irregular bodies on sea bottom scattering, and the use of acoustics in medical physics. In the present study a series of measurements have been taken on a number of solid irregular bodies over a broad frequency range to examine their scattering behavior. A statistical approach has been adopted to measure the ensemble average form function for the irregular bodies and these observations are compared with predictions based on a semiheuristic approach to solving the scattering problem.

10:15

2UW9. Temporal and spatial variation of bottom backscatter in the STRESS experiment. Darrell R. Jackson and J. George Dworski (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

As part of the STRESS (sediment transport events on shelves and slopes) program, bottom backscattering data have been gathered using a circularly scanning sonar operating at 40 kHz. Scans were obtained over a 49-day deployment period at a rate of exactly 10 scans per day. The mechanical and electronic stability of the apparatus made it possible to maintain phase coherence over the entire deployment. As a result, coherent ping-ping correlation can be used to reveal changes in bottom scattering. The results are presented in the form of a false-color movie, which shows that biologically mediated change occurs with several temporal and spatial scales. A point scatterer model has been developed to reduce the measured correlation values to more readily interpreted parameters related to scatterer motion. By fitting the model to data, values for mean-square scatterer displacement between pings are obtained.

10:30

2UW10. Modeling of low-frequency long-range reverberation measured in the CEAREX 89 experiment. T. C. Yang and T. J. Hayward (Acoust. Div., Naval Res. Lab., Washington, DC 20375)

In the Arctic ocean, very low-frequency (10–50 Hz) reverberation returns from the ice and bottom both contribute to the total received reverberation and are not easily distinguishable in long-range reverberation data, except where there is a dominant bottom or ice feature. In this paper, a normal-mode scattering model for surface and bottom protuberance is used to model long-range reverberation data collected during the CEAREX 89 experiment in the Norwegian/Greenland Seas. Modeled reverberation spectrum levels at 23 Hz are compared with data to investigate the relative contributions of the ice and bottom to the measured reverberation. It is found that for a source at 91-m depth in the 3000-m-deep basin, the reverberation level for a receiver at 91 m is dominated by scattering from the ice except for reverberation associated with certain identifiable bottom features. For the same environment but a deeper (244-m) source, reverberation levels from the ice and bottom are more comparable. For a strongly range-dependent environment, returns from bottom features are clearly identifiable in the data.

10:45

2UW11. Nonreciprocity of long-range reverberation in a wedged continental shelf. Ji-xun Zhou, Xue-zhen Zhang (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332), and Dinghua Guan (Inst. of Acoust., Academia Sinica, Beijing 100080, People's Republic of China)

The bottom scattering strength at low frequency and grazing angle in shallow water can only be extracted from long-range reverberation (with large R/N ratio). Due to multipath effects the conventional model of boundary reverberation, based on geometric ray theory in the deep sea, is no longer suitable for calculating the long-range reverberation in shallow water. Up to date little reliable information on bottom scattering at low frequency and small grazing angle has been given in shallow water. For a frequency range of 50–2000 Hz and at a grazing angle of 0.5° – 10° , it is hard to confirm the mechanism of bottom scat-

tering and its dependence on frequency, grazing angle, and bottom type. In such a range, even the angular dependence of bottom scattering (a most basic relationship) is questionable. In this talk, using the WKB approximation to adiabatic normal mode theory, an averaged intensity expression $R(r)$ of long-range reverberation in wedged homogeneous shallow water is derived. For two identical sources directed at each other, the monostatic reverberation intensities, obtained at two terminations with a depth of h_1 or h_2 , would not be reciprocal. It is shown that $R_{h_1}(r)/R_{h_2}(r) = (h_1/h_2)^n$, where n is an angular index of bottom backscattering. Based on this result, the value of n at small grazing angle can be determined from at-sea long-range reverberation data. A simulation tank experiment is suggested. [Work supported by ONR and the IAAS.]

11:00

2UW12. Computational efficiency in a bistatic, range-dependent, normal-mode acoustic reverberation model. David M. Fromm, B. J. Orchard, and Kenneth H. Luther (U. S. Naval Res. Lab., Acoust. Div., Washington, DC 20375-5000)

Acoustic bottom reverberation from a water-sediment interface for bistatic source and receiver configurations in range-dependent, multi-layered, shallow-water environments is calculated with a normal-mode model. The signal field is propagated to and from the scattering regions using adiabatic normal-mode theory. Scattering from the ocean bottom is treated using a three-dimensional Lambert's-law/facet reflection model. An explicit description of the coupling between incident and scattered modes and their contribution to reverberation is obtained. When all possible contributions are considered, the computation time is proportional to the product of the number of incident modes and the number of scattered modes. If the modal contributions to reverberation are weighted and sorted into nonincreasing sequences, then the subsequent summation over all modes may be truncated at a desired level of accuracy. A dramatic increase in computational efficiency may be achieved. Results for a realistic ocean environment are presented which illustrate some of the features of the model. [Work supported by Office of Naval Technology, Code 234.]

11:15

2UW13. Two-dimensional cross correlation of reverberation data. L. Canales, J. Harris (Stanford Univ., Dept. of Geophys., Stanford, CA 94305), E. J. Yoerger, and J. W. Caruthers (NRL, Stennis Space Center, MS 39529)

A technique of time stretching time-series data is applied to reverberation time sequences beamformed from a multielement horizontal receiving array. For different, but nearby, array locations, beams that are scattered off a common area should be highly correlated. The correction method applied here maps each beamformed time series such that common scatterers coincide in position on a 2-D grid. The method uses an average pilot signal and 2-D cross correlations to stretch the time series in a way similar to static corrections in surface seismic processing. This process requires multiple iterations and the average pilot signal is recalculated between iterations. This method was applied to the aforementioned data from five different array locations using an average pilot signal. The results showed a high degree of correlation among the time series. The results are compared with available bathymetry to determine if scattering events can be matched with bottom features. [Work supported by ONR.]

2UW14. Ocean-basin scale inversion of reverberation data. N. C. Makris, J. M. Berkson, W. A. Kuperman, and J. S. Perkins (Naval Res. Lab., Washington, DC 20375)

An optimization method for numerically estimating scattering strength over large areas of the ocean floor is under development. A cost function comparing beamformed towed-array reverberation data with replica reverberation data derived from an ocean-basin scattering strength model is minimized. Data spanning a sufficient set of source locations and/or receiver orientations are inverted simultaneously. Inherent to this process is also the removal of the right-left ambiguity of the beamformed data during the inversion. The method has been successfully applied to synthetic data. It is presently being used to analyze bottom reverberation data acquired during the ONR/SRP Acoustic Reconnaissance Cruise of August 1991. In conjunction, the adiabatic mode formulation [Kuperman *et al.*, *J. Acoust. Soc. Am.* **89**, 125-133 (1991)] is being used to efficiently model acoustic propagation over the variety of source locations required.

2UW15. Reverberation imaging. T. C. Yang (Acoust. Div., Naval Res. Lab., Washington, DC 20375)

Reverberation imaging is a technique that creates an image of surface and bottom scatterers using low-frequency reverberation data. Reverberation from explosive charges of moderate size are known to last tens of minutes covering a basin-wide ocean. The backscattered signals (reverberations) carry information on the nature and locations of boundary features, thus affording wide-area knowledge from a single measurement location. Because of the multipath propagations, conventional processing of backscattered (reverberation) data cannot separate long-range returns arriving simultaneously from the surface and the bottom, except where there is a dominant feature such as a seamount. Matched field/mode processing could in principle be used to localize the surface and bottom scatterers separately except that these methods assume pointlike sources whereas in reality the scatterers are extended objects. This paper addresses modified matched field/mode processing to create images of surface and bottom scatterers separately.

Session 3AO

Acoustical Oceanography: Low-Frequency (< 5 Hz) Sources and Generating Processes in the Sea

Randall L. Jacobson, Chair

Office of Naval Research, Code 1125 GG, 800 North Quincy Street, Arlington, Virginia 22217

Chair's Introduction—1:00

Invited Papers

1:05

3AO1. The SAMSON experiment—Initial results. John A. Orcutt (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., La Jolla, CA 92093)

The ONR sources of ambient microseismic ocean noise (SAMSON) experiment was conducted in October–December 1990 to develop a synoptic view of low-frequency (10 mHz–1 Hz) ocean noise excitation and propagation. In order to understand the effects of ocean-atmosphere coupling, the experiment was conducted in coordination with the ONR surface wave dynamics experiment (SWADE) which provided detailed meteorological and wave dynamics data. The participating universities and laboratories included FlowDrill Industries, the University of Miami, Oregon State University, the Scripps Institution of Oceanography, and Washington State University. The field experiment produced continuous recordings from seismic and hydrophone arrays located on the ocean floor and near the deep ocean surface off the North Carolina coast, at the coastline at Duck, North Carolina, and in the Great Dismal Swamp, inland. During the experiment, a hurricane (Lilly) and a strong Nor'Easter passed through the area causing significant weather, sea surface, and noise pattern changes with frequencies ranging from as low as 5 mHz to more than 2 Hz. The correlations between weather and seismo-acoustic noise will be presented as well as patterns of frequency-dependent evolution of primary noise types including single and double frequency microseisms and pressure fluctuations associated with infragravity waves. Globally distributed earthquakes, which occur at frequent intervals, are responsible for a great deal of noise variability in the frequency band 60–100 mHz.

1:30

3AO2. A hundred-day observation of microseism evolution in shallow water using a 6-point shallow buried OBS/P array. Tokuo Yamamoto and Thomas Nye (Geoacoust. Lab., Div. of Appl. Marine Phys., Univ. of Miami, Miami, FL 33149)

A 6-point shallow buried ocean bottom seismometer and pressure gauge (SPOBS/P) array was deployed 2 km offshore of the Army Corps of Engineers' Field Research Facility at Duck, North Carolina at 15-m water depth as part of the ONR sources of ambient microseismic ocean noise (SAMSON) experiments. An array element consists of three orthogonally mounted accelerometers and a pressure sensor and was buried approximately 1 m below the seafloor. The array aperture was about 1.2 km tuned for shallow-water microseism wavelengths. Although the array aperture is too large for gravity wave directional spectra measurements, the recently developed buried ocean directional spectrometer (BOWDS) method is used to measure the directional spectra of gravity waves simultaneously with the directional spectra of microseisms. The BOWDS method requires only the point measurement of pressure and the two orthogonal components of seabed motion [T. Nye *et al.*, *J. Atmos. Ocean Technol.* 7, 781–791 (1990)]. Over 100-day continuous recording of real-time data was made starting 15 September 1990. The BOWDS results show good agreement with the linear 9-point array of pressure gauges. From this 100-day observation, a consistent pattern of microseism evolution was identified. Microseisms are suddenly activated right after the sudden shift of wind direction as multidirectional seas start to evolve. Fresh microseisms are generated at high frequency, say at 0.5 Hz at the beginning. As the seas get older, the microseisms at lower frequencies are generated. The evolution of microseisms follows the pattern of newly born seas. [Work supported by ONR.]

1:55

3AO3. Veering winds, directionally opposing seas, and double-frequency pressure fluctuations at the sea floor. T. H. C. Herbers and R. T. Guza (Ctr. for Coastal Studies, 0209, Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093)

A 24-element array of pressure transducers (aperture 250×250 m) was deployed on the seafloor in 13-m depth, 2 km offshore of Duck, NC, to investigate the generation mechanisms of double-frequency (0.3–0.7

Hz) pressure fluctuations. Dramatic increases in pressure energy levels at about double the frequencies of locally generated seas consistently occur after sudden large shifts in local wind direction. The observed double-frequency energy levels agree well with predictions based on weakly nonlinear wave theory [K. Hasselmann, *J. Fluid Mech.* **12**, 481–500 (1962)] and the observed frequency-directional spectra of swell and sea, confirming that the double-frequency pressure fluctuations are long-wavelength secondary waves forced by nonlinear interactions between nearly directionally opposing seas. Third-order statistics show the theoretically expected phase-coupling between opposing seas and double-frequency pressure fluctuations. Estimates of wave numbers and propagation directions demonstrate that the double-frequency pressure fluctuations have the theoretically predicted sum vector wave number of the interacting seas.

2:20

3A04. Pressure fluctuations induced by the nonlinear interaction of ocean surface waves. Charles S. Cox and David C. Jacobs (SIO 0230, UCSD, La Jolla, CA 92093-0230)

Pressure fluctuations caused by the nonlinear interaction of ocean surface waves are, in second order, related to sums and differences of the phases of pairs of surface waves. Terms with frequencies above those of the surface waves result from the sum type. When the sea waves include nearly oppositely directed components these pressure fluctuations arrive on the deep seafloor as acoustic waves (horizontal wave numbers $k < \omega/c$), at double the frequency of the generating waves, and can couple with Rayleigh waves in the ocean crust [Longuet-Higgins, *Philos. Trans. R. Soc. London Ser. A* **243**, 1–35 (1950)]. At depths between the sea surface and c/ω the horizontal wave-number distribution is broader. Since each wave-number component decreases exponentially with depth, the integrated pressure intensity increases toward the sea surface, approximately inversely as depth squared. This behavior is to be contrasted with pressure fluctuations induced in the sea by microseismic Rayleigh waves from distant sources. Here, the sea surface acts as a pressure release and the intensity of pressure fluctuations diminishes toward the sea surface. Inferences on the magnitude of local generation of microseisms have been made by measurement of pressure spectra over a range of depths. Local generation was dominant when a storm swell was incident on, and reflected by, a steep, rocky shore. Generation was generally weaker when storm waves were incident on a gently sloping sandy shore. [Work supported by ONR.]

2:45–3:00

Break

3:00

3A05. Relationship of microseisms and ocean wave climatology. Spahr C. Webb (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0205)

Ocean waves have long been recognized as the source of sound in the ocean in the frequency band from 0.05 to 5 Hz. Ocean surface waves from nearly opposing directions interact to excite an elastic wave of much greater phase velocity. Necessarily, the climatology of microseisms is directly related to the climatology of ocean waves. However, the microseism energy propagates as free modes (Rayleigh waves) of the elastic wave waveguide set up by the seafloor and the steep gradient in elastic wave velocities below the seafloor. The propagation of elastic waves within the waveguide is frequency and model dependent. The modes propagate great distances which act to average the ocean wave climate over a large area. The size of this area is frequency dependent and is determined by the e-folding distance associated with attenuation of Rayleigh waves across the ocean. Data from arrays of seismic and pressure instruments in both the Atlantic and the Pacific will be compared to modeling results. The goal is to predict microseism climatology from ocean wave climatology. [Work supported by ONR.]

3:25

3A06. The seismo-acoustic response to the nonlinear interaction of ocean gravity waves. A. C. Kibblewhite and C. Y. Wu (Dept. of Phys., Univ. of Auckland, Auckland, New Zealand)

In the absence of shipping, high sea/wind states can produce measurable noise in the sea, various noise mechanisms dominating in different parts of the spectrum. In the band 1–3 to 50 Hz, the noise source is believed to be breaking waves. Between 0.1 to 1–3 Hz, commonly called the microseism band, the high noise levels observed are believed to arise from nonlinear interactions between surface gravity wave trains. At still lower frequencies (0.02–0.1 Hz), noise levels are generally very low and are thought to be controlled by currents and turbulence in the seafloor boundary layer. In the final band from dc to 0.02 Hz the noise again increases. It is felt that the noise in this band originates from very long wavelength swell. This paper

examines the generation of ocean and seafloor noise in the microseism band, discusses the question of the source level associated with the wave-interaction process, and examines the dependence of the observed seismo-acoustic noise spectrum on wind speed, on seastate (PM and JONSWAP seas), on water depth, and on the geoacoustical environment. Experimental data from onshore seismic sensors will demonstrate the possibility of using the ULF noise field to measure seastate remotely and to provide insight into the processes controlling the growth and decay of wind-driven seas.

3:50

3A07. Scattering of sound into the seafloor waveguide. LeRoy M. Dorman, Anthony E. Schreiner (Marine Phys. Lab., Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0215), and L. D. Bibee (Code 360, Naval Res. Lab.—Stennis, NSTL Station, MS 39529)

Data from seafloor arrays indicate that seafloor noise at frequencies greater than 0.5 Hz propagates as fundamental and higher mode interface waves (Schreiner and Dorman, 1990). For much of this frequency range, the phase velocity of these waves is less than the acoustic velocity in water, so the wave is weakly excited by sources in the water column. Correlations between seismic noise and local wind and swell suggest that, at least at frequencies greater than 0.05 Hz, most of the noise originates at the sea surface, leaving one with the problem of connecting the apparent cause (swell and windwaves) with the effect (seafloor noise). The most plausible explanation is that scattering at or near the seafloor couples energy from the high-velocity modes to the interface waves. Several mechanisms are possible. Scattering at rough surfaces (Kuperman and Schmidt, 1989), volume scattering caused by velocity variations (Dorman *et al.*, 1991), and mode coupling through anelasticity are all candidates. [Work supported by ONR.]

Contributed Paper

4:15

3A08. University of Miami shallow buried OBS experiments at the Oregon Margin. Tokuo Yamamoto and Altan Turgut (Geoacoust. Lab., Div. of Appl. Marine Phys., RSMAS, Univ. of Miami, Miami, FL 33149)

A new method of OBS burial has been successfully tested in 3000-m-deep waters of the Oregon Margin during the cruise on board R/V WECOMA in September 1991. Guralp CME-3 broadband seismometers were buried approximately 1 m below seafloor. Approximately 50 h each of continuous data were successfully collected at OM-1

(44 42.5' N, 125 32.3' W) and OM-8 (45 04.83' N, 125 24.96' W) sites. A local earthquake event at 01:36 a.m., 12 September (Oregon time) was clearly recorded in the seismic channels and pressure channel at OM-1 station. The time-averaged frequency spectra of vertical acceleration and pressure show a quiet notch at 0.04 to 0.10 Hz as well as very high coherency between channels both at the infragravity wave band (0.003 to 0.03 Hz) and at the double frequency microseism band (0.10 to 0.50 Hz). The crustal structures of compressional wave velocity, shear wave velocity, and porosity down to 10 km below seafloor are determined at a resolution of a few hundred meters using BSMP inversion of the infragravity data. [Work supported by NSF.]

4:30-5:30

Panel Discussion

PANEL MODERATOR: Randall L. Jacobson

PANEL MEMBERS: Charles S. Cox
LeRoy M. Dorman
T. H. C. Herbers
A. C. Kibblewhite
John A. Orcutt
Spahr C. Webb
Tokuo Yamamoto

Session 3NS

Noise: Active Noise Control

Jiri Tichy, Chair

Graduate Program in Acoustics, Pennsylvania State University, University Park, Pennsylvania 16802

Chair's Introduction—2:00

Invited Papers

2:05

3NS1. A study of the potential for active control of road noise in automobiles. W. Brent Ferren and Robert J. Bernhard (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

Laboratory experiments were conducted on six vehicle suspension systems to determine the extent to which active noise control could be achieved for random excitation of the tire. In each case, a study was conducted to identify a desirable location for an accelerometer controller input sensor. Control at the driver's head location was achieved using a single-channel commercial, open-loop, digital controller. Broad-band noise reductions between 2 and 11 dB were achieved. The results were compared to analytical predictions which accounted for the limitations of coherence and finite size digital filters. The experimental results were comparable to analytical results when similar digital filters were utilized. Various filter alternatives were also studied analytically. In general, recursive filters performed significantly better than transversal filters. Small improvements were achievable using larger filters than currently available commercially. [This investigation was supported by Nissan Motor Co. and Nelson Industries.]

2:30

3NS2. Active control of radiation from vibrating structures. Scott D. Sommerfeldt (Appl. Res. Lab. and Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

In many applications, one faces the challenge of controlling the sound field that is produced by a vibrating structure. There are two general approaches that can be taken to control such a sound field. The first approach involves directly controlling the acoustic field by means of additional acoustic sources, such as loudspeakers, which are controlled in such a way so as to minimize the noise in the acoustic field. The second approach involves controlling the acoustic field indirectly, in that one can control the structural vibration field in a manner so as to minimize the radiation from the structure. This paper will outline some of the possible methods of using active vibration control of the structure so as to minimize the acoustic radiation from the structure. The physical mechanisms that lead to acoustic radiation and that can be controlled by means of an active vibration control system will be discussed. Research issues related to each active control approach, and the conditions that may make one particular approach for implementing active control preferable over other choices will also be discussed.

2:55

3NS3. Active suppression of radiated sound from a vibrating structure using adaptive modal control. James Thi (AT&T Bell Labs., Arlington, VA 22202) and Dennis R. Morgan (AT&T Bell Labs., Murray Hill, NJ 07974)

Active control for silencing sound and vibration has recently become the subject of much investigation and is now being enhanced by the advent of adaptive filtering techniques. If the active control system is to suppress the sound field at locations where acoustic measurements are available, then the adaptive filtered-x least-mean-square (LMS) algorithm can be used. An alternative approach that does not require acoustic measurements controls the vibrational energy so as to minimize the radiated acoustic field. There are two approaches that can be used to achieve this. The first approach suppresses the total vibrational energy by minimizing an objective function that represents either the sum of the output sensor powers or the sum of all dominant vibrational modal powers. The second approach employs an adaptive modal control system to suppress selected high-efficiency radiating modes from the vibrating structure. In this paper, these two methods are compared and illustrated by numerical example for a simply supported vibrating cantilever beam. Calculations of vibrational energy and acoustic energy with and without control are presented and

3NS4. Stabilization of feedback-type active control systems in finite ducts. David C. Swanson (Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Use of feedback-type control architectures in active noise cancellation can be very difficult due to the extreme sensitivity of this approach to instability, but are sometimes advantageous because the need for the reference sensor is eliminated. Instability results when the feedback loop time delay causes an additional phase shift of 180° at some frequencies allowing high-gain positive feedback to occur. The classical approach to this problem is to design a compensation filter which reduces the feedback gain to less than unity at the frequency(s) where the loop delay causes instability. In a finite-length duct, the feedback instabilities are found to be related to the transducers and the distance between them as well as their positions within the duct. The potential for instability is present whether adaptive or fixed active noise control is done using a feedback-type controller. The compensation filter design is determined directly from the physical parameters of the finite waveguide and the transducers. However, the performance of the feedback controller is restricted by the compensation filter.

Contributed Papers

3:45

3NS5. An impedance model for actively controlled finite waveguides. Ralph T. Muehleisen and David C. Swanson (Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

An analytic impedance model of a finite waveguide with primary and secondary sources is developed utilizing multiple two-port networks. The acoustic source impedance and the waveguide radiation impedance are included in this model. In addition, a control system block diagram is developed for stability analysis of the physical system. Both the analytic model and the control system diagram are used to derive a closed form solution for an optimal cancellation filter for both feedforward and feedback control systems. The effect of radiation and source impedances on the system resonances and both optimal and suboptimal filters are discussed. Suboptimal control filters are important because all adaptive noise cancellation systems must progress through a suboptimal solution before reaching the least-squared error. During convergence additional system resonances can occur, prolonging the convergence time of the adaptive algorithm. Transfer functions in a waveguide are measured and used to design fixed optimal and suboptimal pole-zero feedforward and feedback filters. The steady-state and transient performance of the filters are compared.

4:00

3NS6. Robustness of adaptive active noise control systems. Mark Johnson and George Dodd (Acoust. Res. Ctr., School of Architecture, Univ. of Auckland, Private Bag 92019, Auckland, New Zealand)

In recent years, adaptive control techniques have been increasingly employed in active acoustic noise control systems. The aim in using an adaptive methodology is to produce a noise control system that is entirely self-calibrating and that can maintain optimal attenuation despite changes in the acoustic environment. However, a cost of the increased sophistication of adaptive controllers as compared to fixed control methods is the introduction of some unexpected performance limitations. In particular, active noise control systems based on adaptive methods can suffer from instability, long-term drift and suboptimal steady-state performance. In a recent paper [M. Johnson and G. Dodd, Proc. Internoise 91, Sydney, 1137-1140 (1991)], some of the inherent instability mechanisms in an adaptive active noise control system were identified. The proposed paper will provide both theoretical support for these results and confirmation by way of simulation and real-time examples. It is possible to improve the controller performance without greatly increasing the computational burden by making some straight-

forward modifications to the adaptation algorithm. Results obtained using the modified algorithm demonstrate improved tolerance to disturbances and changing conditions in the acoustic environment.

4:15

3NS7. Multipole sources for cancellation of radiated sound fields. J. Stuart Bolton and Thierry A. Beauvilain (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

It is of interest to be able to cancel the sound field generated by a noise source everywhere exterior to the source. The nature of the secondary source or sources required to achieve that cancellation is a subject of current interest. It has recently been suggested that the sound radiation from coherent, finite-size radiators may be represented as a superposition of monopole fields [G. H. Koopman *et al.*, J. Acoust. Soc. Am. **86**, 2433-2438 (1989)]. It has also been observed that a monopole source may be represented by an infinite-order multipole source placed elsewhere [A. J. Kempton, J. Sound Vib. **48**, 475-483 (1976)]. In principle it is thus possible to create a single multipole source that could represent, and thus cancel, the sound field generated by an arbitrary coherent radiator. In this paper, the initial results of a study to determine the feasibility of such an approach are presented. In particular, the realizable far-field attenuation will be considered as a function of frequency, secondary source order, and primary-secondary source separation distance. It will be shown that useful levels of low-frequency far-field attenuation may be obtained using secondary sources truncated at quadrupole order and positioned within several meters of the primary source. It has thus been concluded that the approach suggested here may find application in the active control of low-frequency exterior sound fields.

4:30

3NS8. Modal control using the RLMS adaptive, recursive algorithm. Christine M. Scheper and Robert J. Bernhard (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

Many current noise controllers use adaptive, digital filters for system identification as an integral part of the controller. For certain applications, it is desirable to use recursive filters in the controller. One of the most efficient adaptive recursive digital filter algorithms is the so-called RLMS algorithm. Previous experience with the RLMS algorithm has shown it works reasonably well for duct active noise applications. How-

ever, performance has been unsatisfactory for applications where the system is described by lightly damped modes. In this investigation, the error surfaces of the modal system are computed and the convergence of the RLMS algorithm along these surfaces is monitored. Light damping and high-sampling frequencies are found to create features in the error surface that slow the convergence of the RLMS algorithm, often to the extent that convergence appears to stop. Control of modal systems using the RLMS algorithm is often limited to very moderate levels compared to levels that are achievable. When the features of the error surface are understood, simple modifications to the algorithm overcome these difficulties.

4:45

3NS9. Dynamic absorbers to control elevator noise in buildings—A case study. Timothy J. Foulkes (Cavanaugh Tocci Associates, Inc., 327F Boston Post Rd., Sudbury, MA 01776)

Dynamic absorbers (tuned mass dampers) were attached at strategic points on a structural beam to cancel a low-level motor tone in the adjacent apartment. The motor and beam were part of an elevator system, and conventional vibration isolation was not feasible. This paper will include vibration measurements, discussion of absorber design, and photographs of the installation.

TUESDAY AFTERNOON, 12 MAY 1992

SALON B, 1:25 TO 5:15 P.M.

Session 3PAa

Physical Acoustics: Nonlinear Physical Systems II

Junru Wu, Chair

Department of Physics, University of Vermont, Burlington, Vermont 05405

Chair's Introduction—1:25

Invited Papers

1:30

3PAa1. Nonlinear acoustics of media with complex structure. I. Yu. Beljaeva, V. E. Nazarov, and L. A. Ostrovsky (Inst. of Appl. Phys., 46 Uljanov St., Nizhny Novgorod, 603600 Russia)

The paper contains a brief review of a recent investigation of nonlinear properties of media with structural inhomogeneities such as pores, grains, cracks, etc. Such media are shown to possess an anomalously strong acoustical nonlinearity (SAN). As examples, the results of theoretical and experimental research of the following media are presented: (1) Rubberlike media [with $\mu \ll \lambda$ (λ, μ are Lamé parameters)] with spherical and cylindrical pores. For such media, the nonlinearity is small. Experiments with a material such as plastizole give a good quantitative agreement with theory. (2) Grainy media (with systems of spherical grains) in which nonlinearity is connected with change of interparticle contact area caused by external pressure. It is established that in this case nonlinear parameters are determined only by the values of initial static deformation. The results of experiments for some samples of crumbly soils give good agreement with theoretical conclusions. The results described seem to be of value in connection with possible new methods of diagnosis in geophysics and seismology.

2:00

3PAa2. Finite amplitude wave studies in earth materials. P. A. Johnson, G. D. Meegan, K. McCall (MS D443, Los Alamos Natl. Lab., Los Alamos, NM 87545), B. P. Bonner (Lawrence Livermore Natl. Lab., Livermore, CA 94550), and T. J. Shankland (Los Alamos Natl. Lab., Los Alamos, NM 87545)

The highly nonlinear elastic behavior of rock may enable new means of interrogating earth structure, of measuring physical properties, and of modeling the seismic source. Compared to uncracked materials, rocks have a large nonlinear response because they contain numerous microcracks that readily compress under applied stress causing large changes of elastic moduli with pressure. Thus nonlinear effects in rocks can be two orders of magnitude greater than those of the uncracked materials typically studied in nonlinear acoustics. Several areas of nonlinear research are currently being undertaken in these laboratories. First, low-frequency (0.1–100 Hz) attenuation studies using a torsional oscillator show that nonlinear coefficients can be greatly increased by inducing additional microcracks in Sierra White granite. Second, ultrasonic parametric array studies demonstrate that strong difference frequency signal generation can take place inside rock samples. Lastly, energy redistribution of finite amplitude waves may produce progressive changes in observed spectra with distance. If a significant amount of energy is redistributed as a function of distance, then source models (based on assuming linear elastic wave propagation) may be in error. This theoretical and experimental work demonstrates that energy redistribution does indeed take place in rock.

3PAa3. Nonlinear effects in periodic and disordered wave media. J. D. Maynard and M. J. McKenna (Penn State Univ., University Park, PA 16802)

While there has been considerable theoretical and experimental research with quantum and classical wave propagation in linear systems, involving effects of Anderson localization, etc., there has been considerably less research with nonlinear media. A fundamental question is whether or not nonlinear effects destroy Anderson localization. There are about eight theoretical papers that treat this question, and half predict that localization is destroyed, and half predict that it is not. An experiment involving nonlinear acoustic wave propagation in a disordered potential field has been performed, and an unambiguous result has been obtained. Also, measurements of nonlinear wave propagation in a periodic field have been made, and in addition to the expected effects of canting and hysteresis, other novel effects are observed. [Work supported by NSF DMR 9000549 and ONR.]

3:00-3:15

Break

Contributed Papers

3:15

3PAa4. Ultrasonic four-wave mixing mediated by a suspension of microspheres in water: Scattering analysis and temporal response. Harry J. Simpson and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Two ultrasonic pump waves are used to produce a grating in a suspension of 25- μm -diam latex particles. A higher frequency ultrasonic wave is used to probe the established grating to produce ultrasonic Bragg scattering. The scattering depends strongly on the pump wave and is an unusual class of nonlinearity. A previously summarized model of the interaction [H. J. Simpson and P. L. Marston, *J. Acoust. Soc. Am.* **90**, 2244(A) (1991)] uses an Epstein layer approximation and effective medium approximation. In the present research, Rayleigh scattering from a sparse layer of spheres is considered in the Born approximation and is found to be consistent with normal reflection from an Epstein layer in the long wavelength limit. This gives a theoretical test of the previous approximation. The temporal response of the suspension and resulting Bragg scattering to sudden changes in the pump amplitude are considered. [Work supported by ONR.]

3:30

3PAa5. Cooperative radiation of acoustic waves by gas bubbles in a liquid. Yu. A. Il'insky^{a)} and E. A. Zabolotskaya^{b)} (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78712-1063)

The process of cooperative radiation of acoustic waves by radial bubble oscillations is investigated theoretically. In addition to instantaneous bubble interaction [E. A. Zabolotskaya, *Sov. Phys. Acoust.* **30**, 365 (1984)], the interaction due to acoustic radiation has also been taken into account. Numerical results for the acoustic intensity produced by ten bubbles that pulsate in water with random initial phases are presented. Mutual bubble interaction causes synchronization of the bubble pulsation phases, which gives rise to collective radiation. The phenomenon is not as strong as in optics. In order to ensure superradiance by ten bubbles, the radial oscillations of the bubbles must have amplitudes of the order of their initial radii. [This work was supported by the David and Lucile Packard Foundation, and by the Office of Naval Research.] ^{a)}On leave from the Department of Physics, Moscow State University, 119899 Moscow, Russia. ^{b)}On leave from the General Physics Institute, Russian Academy of Sciences, 38 Vavilov Street, 117942 Moscow, Russia.

3:45

3PAa6. Nonlinear vibration analysis of an ultrasonic cleaning problem. Thomas L. Geers and Mohamad Hasheminejad (Ctr. for Acoust., Mech. and Mater., Dept. of Mech. Eng., Univ. of Colorado, Boulder, CO 80309-0427)

A micron-size spherical particle is bound to an elastic half-space by van der Waals forces. The particle and the surface of the half-space are immersed in an acoustic fluid and excited by harmonic plane waves propagating normally to the surface from either direction. The vibration of the particle is unidirectional and nonlinear about an equilibrium position determined by van der Waals attraction and Hertz contact. In the manner of a previous linear analysis [T. L. Geers and M. Hasheminejad, *J. Acoust. Soc. Am.* **90**, 3238-3247 (1991)], an equation of motion is formulated that incorporates the effects of particle and fluid inertia, radiation and material damping, and surface and particle compliance. Numerical results are presented for a silicon particle adhering to a silicon half-space both in air and in water. Highly resonant primary response is found, the product of long-wavelength scattering in the resonance-frequency region. This result points to "resonance cleaning" as a promising method for efficient particle removal.

4:00

3PAa7. Characterization of structures in chaotic unsteady flows. Kavitha Chandra, Ayse Kalkan, and Charles Thompson (Dept. of Elec. Eng., Lab. for Advanced Comput., Univ. of Massachusetts—Lowell, Lowell, MA 01854)

This work is aimed at the investigation of techniques for the characterization of chaotic flow. Previous work in this area [C. Thompson *et al.*, *J. Acoust. Soc. Am.* **90**, 2097-2108 (1991)] has shown that an acoustically driven flow in the presence of a spatially varying boundary exhibits a transition from a streaming to a chaotic flow as the amplitude of the acoustic excitation is increased. The analysis of chaos in fluid dynamics is generally carried out in numerical simulations by examining either spatially averaged time signatures, or temporally averaged spatial frames. In the case of unsteady flows, it was found that the randomness in time is characterized by a large variance, and as such cannot be analyzed by time averaged techniques. However, there is an indication that the vortical disturbances evolve in selective regions of a low-dimensional manifold, closely related to the shape and structure of the bounding walls. In this paper, determination of the reduced dimen-

sional spatial trajectory in three-dimensional embedding space along which the vortical disturbances exhibit the highest degree of correlation and hence the estimation of the direction of their evolution will be examined. [Work supported by DOE.]

4:15

3PAa8. Secondary instability of a Stokes boundary layer. Charles Thompson, Vineet Mehta, and Arun Mulpur (Dept. of Elec. Eng., Lab. for Advanced Comput., Univ. of Massachusetts—Lowell, Lowell, MA 01854)

Energy exchange in a Stokes boundary layer between primary unstable disturbances and secondary disturbances having subharmonic spanwise character is discussed. Primary disturbances correspond to the least-stable spanwise mode according to linear theory. By virtue of the Reynolds stress, primary and secondary modes are coupled. Hence, the temporal growth rate of the disturbance field is mediated by the energy balance between primary and secondary disturbances. In this presentation, the energy exchange between these modes and their influence on the structure of the velocity field is discussed. These disturbances are found to be chaotic. Both primary and secondary disturbances follow a period doubling route to chaos. However, the transition amplitude of the acoustic modulation for which chaos occurs is a function of the wave-number content of the initial disturbance field. The energy exchange between spanwise modes with respect to the initial wave-number content and the modulation amplitude is discussed. [Work supported by DOE.]

4:30

3PAa9. Spatial evolution of unstable disturbances in a Stokes boundary layer. Charles Thompson and Vineet Mehta (Dept. of Elec. Eng., Lab. for Advanced Comput., Univ. of Massachusetts—Lowell, Lowell, MA 01854)

A model describing the process of transition to chaotic fluid motion in an acoustical field driven in time is presented. It is shown that the unstable disturbances occur locally in space. The region of space where instability ensues is given in terms of a local amplitude conditioned on the basic flow. Subcritical behavior of the energy amplitude is shown to occur near the turning point. The chaotic evolution of disturbances in the streamwise direction is studied. Consideration is also given to the interaction of spanwise Fourier modes. Results for the spatial and temporal structure of the disturbance are given. The nature of the chaotic behavior is described in terms of physical parameters. [Work supported by DOE.]

4:45

3PAa10. Propagation of finite amplitude sound in a waveguide with a parabolic sound velocity profile. Douglas E. Reckamp (Appl. Res. Lab., Univ. of Texas at Austin, Austin, TX 78713-8029), Evgenia A. Zabolotskaya, and Mark F. Hamilton (Univ. of Texas at Austin, Austin, TX 78712-1063)

The propagation of finite amplitude sound in a waveguide with a parabolic sound velocity profile is investigated theoretically. It is assumed that the primary wave propagates in a single mode at a frequency that is large compared with the cutoff frequency. The acoustic energy is therefore concentrated near the axis of the sound channel. Both the primary wave and the nonlinearly generated second harmonic component have mode shapes that are described by Gauss-Hermite eigenfunctions. For a primary wave in mode m , the second harmonic component is generated in all even-order modes up to $2m$, with most of the energy contained in mode $2m$. A nonlinear Schrödinger equation is derived for the envelope of a narrow-band pulse that propagates in a single mode. Analytical expressions are derived for the coefficients in the Schrödinger equation, and it is found that only dark envelope solitons can propagate without distortion in the waveguide. [DER was supported by the ARL Naval Academy Scholarship, EAZ and MFH by the David and Lucile Packard Foundation and by the Office of Naval Research.]

5:00

3PAa11. On the possibility of delaying shock formation in finite amplitude sound beams through alteration of the source signal. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332) and Gee-Pin Too (Natl. Cheng Kung Univ., Tainan, Taiwan, ROC 70101)

It was pointed out by Rudnick [J. Acoust. Soc. Am. 30, 564-567 (1958)] that if a plane wave is generated in the form of a backward sloping wave through appropriate phasing of the higher harmonics, then the amplitude of the fundamental frequency received at a given location would be enhanced. Burns [J. Acoust. Soc. Am. 45, 210-213 (1969)] further noted that such a wave would form a shock at a farther distance than that for monochromatic input at the level of the fundamental. Rogers [private communication (1987)] suggested that this effect would be especially beneficial for a sound beam. If the region for shock formation could be moved from the near field, where the signal is quasiplanar, to the far field, where it propagates in a spherical manner, then the result would be a substantial enhancement in the shock formation distance, because of the slower rate at which nonlinear effects accumulate in the far field. The present work uses the NPE code modified for sound beams to test this possibility. The input for the analysis is a baffled piston whose vibration consists of a fundamental at fixed amplitude plus a second harmonic at arbitrary amplitude and relative phase. Results for the fundamental and second harmonic at various locations indicate that the improvement is negligible because of the relative phase shift associated with transition from near-field to far-field behavior.

Session 3PAb

Physical Acoustics: General Topics

Ronald A. Roy, Chair

Applied Physics Laboratory, University of Washington, Seattle, Washington 98034

Contributed Papers

1:30

3PAb1. Parametric ultrasound heating: Theory and modeling. Kenneth D. Rolt and Henrik Schmidt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Recently, the nonlinear effects of focused ultrasound have been studied to evaluate increased absorption and enhanced heating. These features must be established for safety's sake in certain ultrasound imaging and hyperthermia applications. One version of nonlinear acoustics in ultrasound that has not been investigated at length is the parametric application, where separate transducers (or arrays) have coincident, simultaneous foci. This forestalls the mutual nonlinear interaction in the near fields and establishes it only in the confocal region; this is worthwhile because this region is where the heating is supposed to occur. Parametric sonars have been known since the 1960s in the transducer, ocean, and nonlinear acoustics fields but have not been "reinvented" in ultrasound. This paper will cast parametric sound in terms of intersecting focused ultrasound, and will show the theory and the results of a brute-force finite difference model which incorporates linear and nonlinear propagation, and heat transfer.

1:45

3PAb2. Parametric ultrasound heating: Experiments. Kenneth D. Rolt (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

In companion to the preceding talk, this paper will describe the design and results of two experiments in parametric ultrasound heating. Two focused transducers were operated at 0.9 and 1.5 MHz, respectively, and had coincident foci incident upon a 0.08-mm-diam Chromel-Constantan thermocouple. The thermocouple was embedded in an encapsulate having absorption properties, sound speed, and density on-par with human thigh tissue. For these tests, the results show a temperature nonlinearity factor $\sigma_{\text{heat}} = 1.05$, where $\sigma_{\text{heat}} = T_{a+b}/(T_a + T_b)$, T_{a+b} is the temperature rise for simultaneous operation, and $T_a + T_b$ is superposition of the temperature rise for separate operation. This nominally equates to a 1 °C temperature increase for the same electrical input power when both transducers operate simultaneously than when operated separately.

2:00

3PAb3. Experimental study of temperature elevation in a tissue phantom generated by an operating ultrasonic transducer. Joseph D. Chase and Junru Wu (Dept. of Phys., Univ. of Vermont, Burlington, VT 05405)

The temperature elevation in a homogeneous tissue-mimicking material generated by an ultrasonic transducer (focusing or nonfocusing) operating in megahertz frequencies was measured. It was found that in the vicinity of a transducer front face, both transducer surface heating

and ultrasound absorption are important sources for tissue heating. As for heating due to ultrasound absorption, the experimental results are compared with theoretical predictions for nonperfused tissue resulting from a Gaussian beam model and from the model used by the NCRP (National Council on Radiation and Measurements).

2:15

3PAb4. A noninvasive method for measuring muscle filament stiffness using a calibrated glass microneedle. Markku Oksanen (Dept. of Phys., Univ. of Helsinki, Siltavuorenpenger 20D, SF00170 Helsinki, Finland), David M. Warshaw, and Junru Wu (Univ. of Vermont, Burlington, VT 05405)

Muscle is believed to contract by the relative sliding of actin and myosin filaments as ATP is hydrolyzed. In order to measure the force generated by myosin during actin filament sliding an *in vitro* motility assay has been developed [A. Ishijima *et al.*, *Nature* **352**, 301–306 (1991)] where one end of a single actin filament is attached to a thin glass microneedle and the other end is brought into contact with a myosin-coated coverslip in the presence of ATP. As the myosin tugs on the actin filament, bending of the microneedle occurs and allows the force to be calculated provided its stiffness is known. Recently, the stiffness of a microneedle as small as 0.1 pN/nm was determined in this lab by measuring the Brownian motion of the microneedle in air using a fiber-optic Michelson interferometer and the equipartition theorem. The results of this noninvasive method will be compared with those resulting from other invasive methods. [Work supported by NIH HL45161 and BF Goodrich Simmonds Precision Aircraft System.]

2:30

3PAb5. Fractionation of fine particle suspensions by coordinated planar ultrasonic and laminar flow fields in a rectangular acoustic chamber. Z. I. Mandralis and D. L. Feke (Dept. of Chem. Eng., Case Western Reserve Univ., Cleveland, OH 44106)

Acoustics fields are being used to manipulate fine particles (0.1–20 μm) in liquid suspensions held within a thin rectangular channel. Ultrasonic fields are applied across the channel, resulting in particle migration in this direction. Particles with different physical characteristics (size, density, compressibility, shape) respond with different speeds to the sound field [Z. I. Mandralis and D. L. Feke, *Fluid/Part. Sep. J.* **3**, 115–121 (1990)], resulting in partial separation across the channel. These partial separations are amplified into useful separations by applying cyclic, coordinated, bidirectional laminar flows. The response of the particles to the standing wave field will be described. A mathematical model that simulates the fractionation scheme has been developed. Results from the simulation demonstrate the effectiveness of the technique. Experimental results from a batch fractionation of 325 mesh polystyrene particles are presented. Scaleup and continuous operation schemes will be discussed.

3:00

3PAb6. A quasilinear theory of flexible-walled cavity resonators. T. Douglas Mast (Graduate Prog. in Acoust., Penn State Univ., 117 Appl. Sci. Bldg., University Park, PA 16802)

A flexible-walled cavity can be excited into a Helmholtz-like resonance by shearing flow of an incompressible fluid. The mass element is associated with the fluid flowing through the cavity opening; the compliance element is due to the cavity walls. The excitation mechanism involves feedback between the pressure oscillations in the cavity and the unsteady flow in the opening. In this paper, an analytical quasilinear theory of this system is presented. The linear instability of the shear layer, vortex shedding from the upstream edge of the cavity, and structural nonlinearity of the cavity walls are considered. Application is made to the resonance of aneurysms in the bloodstream, incorporating aneurysms' measured elastic properties and other characteristics of the cardiovascular system. [Work supported by the William E. Leonhard Endowment to the Penn State Univ.]

3:15

3PAb7. Studies of ultrasonic velocity, internal friction, and dielectric behavior of aluminum-doped zinc manganese mixed ferrites. Janjanam S. Ramamurthy, N. Veerayya, and K. V. V. M. Mohana Rao (Dept. of Phys., AUPG Ctr., Nuzvid-521201 AP, India)

Ultrasonic velocity, Young's modulus, and internal friction have been studied in aluminum-doped zinc manganese ferrites as a function of aluminum concentration and temperature (300–600 K). The presence of aluminum impurities is found to significantly alter the ultrasonic properties and also the internal friction. In order to ascertain if a similar dependence is reflected in the dielectric behavior as well, the dielectric constant and loss tangent have also been studied in the same ferrites as a function of dopant concentration, temperature, and frequency of measurement. Relaxational behavior is noticed with a remarkable dependence on aluminum impurity concentrations. Results of ultrasonic and dielectric measurements have been explained in the light of cationic exchanges between octahedral and tetrahedral sublattices of the ferrite.

3:30

3PAb8. Transient acoustic wave modeling with higher-order WKBJ asymptotic expansions and symbolic manipulation. Martin D. Verweij (Lab. of Electromag. Res., Dept. of Elec. Eng., Delft Univ. of Technol., P. O. Box 5031, 2600 GA Delft, The Netherlands)

A combination of the WKBJ asymptotic expansion and the Cagniard-De Hoop method is used to determine the *total* space-time domain acoustic wave field in a continuously layered configuration. In the transform domain, a recurrence scheme for the coefficients of the WKBJ asymptotic expansion is derived from the integral equations of the problem. This scheme is well suited for implementation in a symbolic manipulation program, e.g., *Mathematica*. The WKBJ asymptotic expansion is transformed back to the space-time domain by applying the Cagniard-De Hoop method in a particularly efficient way. For various configurations numerical results are presented, which show that the presented approach yields accurate results for instants ranging up to many times the arrival time.

3:45

3PAb9. Physics of standing waves in circular underexpanded jets. Alan Powell (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792)

Although the quasiperiodic structure of ideal, slightly underexpanded, round supersonic jets is "well-known" mathematically (Prandtl, 1904; Pack, 1950), its physical nature has lacked some interpretation. The "diamond pattern" is overt in schlieren photographs, the oblique (conical) waves transversing the jet once in distance $L = 2R\sqrt{M^2 - 1}$ (M is the Mach number; R is the jet radius). The jet returns to the initial condition of uniform perturbed pressure with reversed sign and slightly reduced amplitude, after distance $2L$, with the same sign at $4L$, repeating at $4L$ intervals. The contrast with the two-dimensional case (corresponding distances L and $2L$) is attributable to $\pi/4$ phase change at the axial focus: The initial step wave emerges as an impulselike wave that has little effect, *per se*, on the boundary shape when reflected there, becoming a step wave again after the next focus. However, the cell structure has period $s = 1.640R\sqrt{M^2 - 1}$, corresponding to minima, very nearly, of the jet diameter. Contrasting to square jets, these minima are apparently not associated with any discrete event. [Work supported by the Texas Advanced Research Program.]

4:00

3PAb10. Observations on subsonic edge tones. Alan Powell and Noel Mascarenhas (Dept. of Mech. Eng., Univ. of Houston, Houston, TX 77204-4792)

Discrete tones, sometimes multiple, occur for turbulent jets with exit Mach numbers (M) between 0.14 and 1.0 at nozzle-to-edge distances of $3 < h/d < 55$ ($d = 1$ mm), with an 8° wedge. The minimum h/d for discrete tones to occur increases linearly with Mach number for $0.14 < M < 1.0$. Broadband tones occur at h/d below this minimum distance. As in the classical edge tone, the frequency of the strongest tone increases proportionally with exit velocity at a constant h/d and is approximately inversely proportional to $h/d(f \sim h/d^{-1.1})$ at a constant exit velocity. Schlieren photographs reveal the asymmetric nature of the jet instabilities and the out-of-phase sound waves on either side of the jet. Up to three continuous stages occur, with frequency jumps between them. U_{con} (convection velocity of the jet instability) varies from $0.22 U_{exit}$ to $0.45 U_{exit}$ (U_{exit} = fully expanded jet exit velocity) for $0.22 < M < 1.0$ and $0.12 < M_{con} < 0.45$, $M_{con} = U_{con}/c_a$ (c_a = sound speed). The sound pressure level (SPL) peaks within $1 < h/d < 5$ of the minimum h/d . [Work supported by the Texas Advanced Research Program.]

4:15

3PAb11. Real lift versus acoustic lift: A unifying spectral representation of the basic force/velocity kernel for the entire linear subsonic regime. R. Martinez (Cambridge Acoust. Associates, Inc., 80 Sherman St., Cambridge, MA 02140)

This paper considers an elemental harmonic force in a uniform freestream U . It uses a wave-number transform to express the basic transfer function linking this differential lift to its induced fluid velocities. The analysis shows that the momentum equation for the convective medium introduces a pole in the computed spectrum, and that this pole in turn generates a shed wake for points downstream of the lifting

element. A fundamental benchmark limit for the formulation is the condition of pure acoustic diffraction, obtained from letting $U \rightarrow 0$: the wake pole then disappears, as it should, and the kernel displays its well-known "highly singular" self-effect of $O(\bar{x} - \bar{x}_0)^{-2}$, where \bar{x}_0 and \bar{x} are source and receiver points. The paper's main conclusion is that the

shed wake predicted for an extended lifting surface has a differential cause, not just a global one, and that standard aerodynamic analyses have apparently hitherto concealed this "textured" feature of the unsteady flow field by prescribing a dynamic "macro" constraint on the bound circulation of the aeroscattering problem.

TUESDAY AFTERNOON, 12 MAY 1992

SALON F, 1:30 TO 5:05 P.M.

Session 3PP

Psychological and Physiological Acoustics: Memorial Session for Eberhard Zwicker

Søren Buus, Chair

Department of Electrical and Computer Engineering, 409 Dana Research Building, Northeastern University, Boston, Massachusetts 02115

Chair's Introduction—1:30

Invited Papers

1:35

3PP1. Eberhard Zwicker: The man behind the scientist. Ulrich Tilmann Zwicker (Feldstr. 5, D-8012 Icking, Germany)

In order to fully comprehend and appreciate the scientist Eberhard Zwicker, one has to look deeper and try to see the complete colorful personality that he was. Apart from his involvement in science and engineering, he was a dedicated family man, a loving and challenging husband, a wonderful father. He was an avid and very successful gardener and hobby farmer, an accomplished craftsman with a particular passion for working with wood. He was very partial to extended hikes through the Bavarian and Austrian Alps, as well as to taking a thorough look into the heavens as a hobby astronomer, and incessantly dedicated in his boundless love, respect, and interest for all of creation. It was this interaction with nature, and with beings and things in their natural environment, that led him to a particular, generalistic, almost ecological approach in his area of science. Not only was he interested in isolated academic phenomena, but he wanted to know about the general relations with other phenomena involved. Although he was devoted to basic research, which he thought of as irreplaceable, he needed to see practical applications of this research, in particular applications directly beneficial to humankind. Whereas he wanted to see and learn about many things, he also found it most important to see the many sides of those things, the connections, the overall pattern. This wholesome approach, the joy of which he also desired to share via his teaching, is the basis for Eberhard Zwicker's work, and taking on this approach will give the student of his work much improved insight.

1:55

3PP2. Variations on E. Zwicker's theme of frequency discrimination. Jozef J. Zwislocki (Inst. for Sensory Res., Syracuse Univ., Syracuse, NY 13244)

Eberhard Zwicker once suggested that human frequency discrimination was controlled by the apical cutoff of cochlear excitation rather than by the maximum of excitation [E. Zwicker and R. Feldtkeller, *Das Ohr als Nachrichtenempfänger* (Hirzel, Stuttgart, 1967)]. Physiological evidence supporting Zwicker's suggestion has been obtained. Microelectrode recordings from the cochlear inner and outer hair cells have revealed that the sound frequency associated with their maximum excitation at a given cochlear location changes with SPL, whereas the subjective pitch remains nearly constant. In Mongolian gerbils, at the 2-kHz cochlear location, the frequency is lowered by about an octave when the SPL is increased from 20 to 80 dB [J. J. Zwislocki, *Acta Otolaryngol.* **111**, 256–262 (1991)]. Over the same SPL range, the human pitch changes by only a few percent. A similar near invariance in the high-frequency cutoff of the hair-cell excitation has been found. As a result of the known spatial frequency mapping in the cochlea, this cutoff corresponds to the apical cutoff of hair-cell excitation. The results suggest that the cutoff rather than the excitation maximum constitutes the place code for pitch, especially since the excitation maximum is not sharp enough to account for the great sensitivity of humans and animals to frequency changes. This is consistent with Zwicker's hypothesis. [Work supported by NIDCD.]

3PP3. The number and nature of OAE generators. Susan J. Norton (Children's Hospital and Med. Ctr., 4800 Sand Point Way NE, Seattle, WA 98105 and Department of Otolaryngology-Head and Neck Surgery, Univ. of Washington, Seattle, WA) and Lisa J. Stover (Boys Town Natl. Res. Hospital, Omaha, NE)

Professor Zwicker immediately appreciated the importance of David Kemp's discovery of otoacoustic emissions (OAE), and devoted much of his efforts in recent years to their study. OAEs are sounds generated by nonlinear, highly tuned, physiologically vulnerable biomechanical elements within the cochlea. They can be measured in the external ear canal, and thus provide an exquisite noninvasive tool for studying cochlear mechanics, particularly the "cochlear amplifier," *in vivo*. There are several OAE phenomena, distinguished primarily by the stimuli used to evoke them. While all OAEs are generally absent if there is a cochlear hearing loss involving absence of outer hair cells, how these phenomena relate to one another and normal cochlear function is not entirely clear. For a given evoking stimulus, OAEs differ in their "thresholds," growth rate and maximum output across audiologically normal ears. In addition, within an ear there are frequencies and presumably cochlear regions with relatively stronger emissions than others. Similarly, within an ear different OAE phenomena in the same frequency region may have different thresholds, growth rates, and maximum output. Such differences within and across ears and OAE phenomena may reflect differences in generators or may be the result of procedural differences which require some discussion before more esoteric issues can be addressed. [Work supported by NIDCD and DRF.]

3PP4. Frequency selectivity, critical bands, and focused attention. Bertram Scharf (Auditory Perception Lab., Northeastern Univ., Boston, MA 02115 and CNRS LMA, 13402 Marseille, France)

Zwicker made evident the ubiquity of the critical band in auditory processing, from the detection of tones in noise to the discrimination of amplitude from frequency modulation. Recently, it has become clear that the critical band defines also the limits of the attention band; listeners who focus on a given frequency detect tones at that frequency and at frequencies in the surrounding critical band but miss tones farther away in frequency. The formation of the attention band in the appropriate frequency region may be effected by efferent signals to the cochlea. Support for this hypothesis is based on preliminary data from patients whose olivary-cochlear bundle had been severed to relieve severe vertigo and who reveal greatly widened attention bands. [Research supported by NIH.]

3PP5. Temporal resolution within the upper "accessory excitation" region for maskers with fluctuating envelopes. David A. Nelson (Dept. of Otolaryngol., Univ. of Minnesota, 2630 University Ave. SE, Minneapolis, MN 55414)

During Zwicker's numerous investigations of the time structure of masking, he observed that envelope masking period patterns were larger for frequencies above the masker (in the upper "accessory excitation" region) than for frequencies near the masker (in the "main excitation" region). For example, the peak-to-trough ratio of the masking period pattern for a 4-Hz 100% amplitude-modulated masker was only 12 dB when the probe tone was within the main excitation region, at 1000 Hz, while the ratio was as large as 36 dB when the probe tone was within the accessory excitation region, at 3200 Hz [E. Zwicker, *Acustica* 36, 113-120 (1976)]. In the present research, masking at the peak and within the trough of 500-Hz amplitude-modulated maskers was investigated as a function of test frequency and masker level for modulation frequencies from 4 to 128 Hz in normal-hearing and hearing-impaired ears. Peak and trough masked thresholds were measured using amplitude-modulated signals with phasic and antiphase envelopes. The results from normal-hearing ears confirm Zwicker's finding that temporal resolution can be up to three times better in the upper accessory excitation region than in the main excitation region. Masking during envelope peaks remains constant while masking during envelope troughs increases with modulation frequency. This finding, and the slopes of the growth of masking, support Zwicker's interpretation that masking during the envelope peaks and troughs are associated with simultaneous and nonsimultaneous masking, respectively. Peak-to-trough ratios in the upper accessory region are considerably reduced in hearing-impaired ears, which has important implications for processing speech signals in the presence of maskers with fluctuating envelopes. [Work supported by NIDCD.]

3PP6. Information-bearing features of complex sounds. Ernst Terhardt (Electroacoust. and Audiocommun., Tech. Univ., D-8000 München, Germany)

While Eberhard Zwicker's psychoacoustic and physiological work was mostly of the fundamental research type, he was extremely concerned about employment of insights and models to human benefit, i.e., hearing diagnostics, hearing aids, cochlear implants, tactile hearing prostheses, aids for the blind, and automatic recognition of speech. These types of application have in common that they are crucially dependent on reduction of complex auditory signals to information-bearing features. In fact, auditory acquisition and processing of information was a prominent guideline of Zwicker's scientific endeavor, in which the present author through almost three decades had the privilege to participate. As a result, there emerges a picture of auditory information processing that includes significant insights into the perception of complex sounds such as speech and music. While it is *a priori* evident that auditory information processing is dependent on hierarchically organized, knowledge-based decision mechanisms, the finding is remarkable that pertinent processes are to a considerable extent installed on low levels of the auditory system. Pitch turns out to play a prominent role as a carrier of information on external acoustic objects. These findings throw a new light on early ideas on the significance of auditory frequency analysis in the cochlea, of lateral inhibition, and of interaural pitch formation.

4:15

3PP7. Search for Zwicker's specific loudness function. Rhona Hellman (Auditory Perception Lab., Northeastern Univ., Boston, MA 02115 and VA Outpatient Clinic, Boston, MA 02114) and Mary Florentine (Northeastern Univ., Boston, MA 02115)

A key ingredient of Zwicker's loudness model is the "specific loudness" function which relates loudness to excitation within a single critical band or Bark. For narrow-band stimuli near 1 kHz this relation is described by a power function with a mid-to-high level slope (exponent) about 20% lower than the overall slope of the loudness function. In normal hearing, a slope reduction can be obtained by partially masking a tone with an adjacent high-frequency band of noise. Implicit in these findings is the assumption that the noise effectively contains the high-frequency excitation evoked by the tone. Recent loudness data obtained by magnitude scaling procedures for 21 listeners with bilateral sloping high-frequency losses provide a validation of this assumption. Measured at a frequency beyond which earphone corrected thresholds increase by more than 50 dB/oct, loudness increases with level at a slower rate than at a lower frequency and in normal hearing. Shallower threshold functions, like high tone-to-noise ratios, have little, if any, effect on the tone's rate of loudness growth. The overall shape and slope of the loudness functions can be predicted from Zwicker's model of loudness summation modified to account for noise-induced hearing loss [M. Florentine and E. Zwicker, *Hear. Res.* 1 (1979)]. [Work supported by the VA Rehab. R&D Service and by NIDCD.]

4:40

3PP8. Subjective and physical evaluation of noise immissions. Hugo Fastl (Inst. of Electroacoust., Tech. Univ. München, 8000 München 2, Germany)

For measurements of noise immissions, presently the equivalent level L_{eq} is used. L_{eq} represents the level of a stationary continuous sound with the same sound energy as produced by a noise immission with time-variant level. Hence, the L_{eq} concept is based merely on a physical magnitude, i.e., sound energy, and lacks a solid psychoacoustic background. Therefore, together with Eberhard Zwicker, a method to evaluate noise immissions subjectively in psychoacoustic experiments was implemented. In short, the perceived loudness of a noise immission is represented by the length of a line displayed on a PC monitor. Examples of the subjective evaluation of noise immission with this method are given for road traffic noise and aircraft noise. In addition, the subjective evaluations are compared to physical measurements by a sound level meter and by Zwicker's loudness meter. [Work supported by DFG, SFB 204.]

Session 3SA

Structural Acoustics and Vibration: Condition Monitoring and Diagnostics as an Emerging Technology

Aynur Unal, Chair

Vibration and Sound Research Institute, 1625 Alameda, San Jose, California 95126

Chair's Introduction—2:00

Invited Papers

2:05

3SA1. Monitoring microdrilling operations with an intelligent diagnostic system. I. N. Tansel (Mech. Eng. Dept., Florida Internatl. Univ., Miami, FL 33199)

Microdrills have a short and unpredictable tool life. It is also very difficult to monitor the condition of the microdrill tip with the unaided eye. In this paper, a new approach is proposed to detect failure of the microdrill with a 0.39-mm diameter. The proposed method measures the thrust force by using a dynamometer and tool displacements with a laser vibrometer. The characteristic features of the data are obtained by modeling the profile of the cutting forces measured by the dynamometer and large variations of the laser vibrometer signals. The encoded data are interpreted to estimate tool condition by using the adaptive resonance theory (ART2)-based neural networks. The proposed approach was tested by using microdrills with a 0.39-mm diameter. After testing, less than 10% error was observed during identification of normal and poor operation conditions (just before breakage, breakage, and drilling with a broken tool) with proper vigilance values.

2:30

3SA2. Application of wavelet transforms for characterizing ultrasonic signals. Satish S. Udpa (Dept. of Elec. Eng., Iowa State Univ., Ames, IA 50011)

Ultrasonic methods of nondestructive testing (NDT) are used extensively in many applications ranging from detection of cracks in nuclear power plants to monitoring of wear in jet engines. A crucial step in NDT is the solution to the inverse problem, where the ultrasonic reflections are analyzed for extracting information concerning the shape, size, and location of defects. A multiresolution signal analysis technique, based on the wavelet transform of the ultrasonic signals, is described. The basis function or wavelets used in the analysis are generated from a single prototype by dilation/contraction and shifts. This provides the multi-scale analysis required in the processing of nonstationary signals. The wavelet transform representation of the ultrasonic waveform is then classified employing a pattern recognition algorithm for characterizing the underlying defect. The effectiveness of the method was verified using a set of experimental signals obtained during inspection of power plant tubing. [Work reported is supported by EPRI under Contract RP 3148-06.]

2:55

3SA3. A novel approach to the solution of inverse problems. Lalita Udpa (Dept. of Elec. Eng., Iowa State Univ., Ames, IA 50011)

Solution to inverse problems is of interest in many fields of science and engineering. As an example, in the field of nondestructive testing, such solutions offer a tool for automatic defect characterization in materials from transducer measurements. Inverse problems are frequently described by Fredholm integral equations of the form, $\int_a^b k(x,y)z(y)dy = u(x)$ ($c < x < d$), where $u(x)$ represents the measured data, $z(y)$ represents the source function or system parameters, and $k(x,y)$ represents the kernel of the transformation. The objective is then to solve for the source function from known measurements. This problem is inherently sensitive to the system parameters z , to the shape of the kernel k , and to the accuracy of the measurements u . The paper presents a novel strategy for solving Fredholm integral equations using Hopfield type neural networks. The major advantage of this method is the stability of the solution that comes from the high degree of parallelism and interconnectivity encountered in the neural networks. The algorithm consists of formulating the problem in the form of an error minimization equation. By comparing the error function to the energy function of the Hopfield neural network, the circuit parameters of the network are estimated. Simulation results demonstrating the effectiveness of the approach for obtaining the solution to integral equations are presented.

3:20

3SA4. Intelligent signal processing for source location in anisotropic materials. Karthikeyan Chittayil (Dept. of Industrial and Management Systems Eng., Penn State Univ., University Park, PA 16802)

Active as well as passive source location methods are commonly used in industry, in nondestructive testing (NDT) of materials and structures. Most of these methods cater to materials or media that are isotropic; methods for source location in anisotropic media are the subject of a few recent investigations [Castagnede *et al.*, *J. Acoust. Soc. Am.* **86**, 1161 (1989); Steinberg *et al.*, *J. Acoust. Soc. Am.* **90**, 2081 (1991)]. In the present work, a method of source location in anisotropic media is introduced, which uses the learning ability of a neural-networklike computational algorithm to model the source location problem. During the learning phase, the correlation of the source location with the multisensor data is determined and stored in a "memory matrix." This matrix is then used for the prediction of the location of any unknown source using multisensor input data, by the method of autoassociative recall. The results of the application of this algorithm, to a specific problem of estimating the location of a source in a composite material plate, is presented and discussed.

3:35

3SA5. Fault detection in helicopter gearboxes using neural networks. Michael R. Dellomo (MITRE Corp., M. S. W622, 7525 Coleshire Dr., McLean, VA 22102)

One of the most dangerous problems that can occur in both military and civilian helicopters is the failure of the main gearbox. Currently, the principal method of controlling gearbox failure is to regularly overhaul the complete system. This presentation considers the feasibility of using a neural network to perform fault detection on the accelerometer data. The details and results obtained from studying the neural network approach are presented. Some of the underlying physics will be discussed along with the preprocessing necessary for analysis. Several networks were investigated for detection and classification of the gearbox faults. The performance of each network will be presented. Finally, the network weights will be related back to the underlying physics of the problem.

3:50

3SA6. Sensitivity analysis for acoustic conditioning monitoring of beams. Xiu-Ting Cheng, Robert D. Finch (Dept. of Mech. Eng., Univ. of Houston, 4800 Calhoun Rd., Houston, TX 77204-4792), Pierre Y. Robin, and Ben H. Jansen (Dept. of Elect. Eng., Univ. of Houston, 4800 Calhoun Rd., Houston, TX 77204-4793)

Beams have been used to study the sensitivity of the vibration monitoring method of flaw detection. Cracks and other defects can cause shifting and splitting of resonant frequencies, so that these changes can be used to indicate the presence of the defect. The limitation of the vibration monitoring method is that other conditions, such as uncertainties in the geometry of the test object, its surface conditions, and loading, can also affect the vibration response, and it is necessary to

distinguish the effects due to harmful conditions from those due to harmless ones. The sensitivity of the method is thus determined by the need to make this distinction. These concepts will be illustrated with experimental results from a test fixture for the case of a slotted beam, assuming that a slot can be used as a model for a crack. The effect of geometrical uncertainties, coatings of damping material, and loading can also be studied experimentally. It is possible to use Thomson's theory to calculate the amounts of frequency changes due to a slot, so that the experimental results will be discussed in the light of this theory. [Work supported by NSF Grant No. MSS-9024224.]

4:05

3SA7. Acoustic monitoring of beams by time-frequency analysis. Pierre Y. Robin, Ben H. Jansen (Dept. of Elec. Eng., Univ. of Houston, 4800 Calhoun Rd., Houston, TX 77024-4793), Xiu-Ting Cheng, and Robert D. Finch (Dept. of Mech. Eng., Univ. of Houston)

Acoustic vibrations can be used to monitor the structural integrity of metallic beams. A current technique involves impacting a beam with a hammer and analyzing the induced vibrations. However, the generated vibrations exhibit transient and nonstationary characteristics such as rapid decay of amplitude and time-dependent frequency content. This observation limits the use of classical spectral analysis which requires stationarity. Time-frequency representations, such as the Wigner distribution, provide a more general framework to analyze transient and nonstationary signals. Actual beam vibrations, analyzed with time-frequency representations, will be presented. In particular, it will be shown that this analysis enhances the appearance and disappearance with time of specific frequency components as well as changes of the resonant frequencies. These observations can ultimately be used as discriminative features in a pattern recognition scheme classifying good and cracked beams. [Work supported by NSF Grant No. MSS-9024224.]

4:20

3SA8. Estimation of position of a crack in a concrete block by vibration analysis. Masato Abe, Kiyohito Fujii, and Toshio Sone (Comput. Ctr., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai, 980 Japan)

This paper describes a method to estimate the position of a crack in a concrete block using many sensors. An array of sensors is attached on the concrete block, and a vibration pulse is forced by using a small hammer. If there is a crack, a reflection wave is generated from the position of the crack. Therefore, conventional methods to estimate the position of the sound source seem to be useful for this purpose. However, since the concrete block is elastic, there are several vibration modes, and the position of a crack cannot be estimated by the conventional methods. Therefore, a new method is proposed to estimate the position of a crack in an elastic block. This is achieved by designing a FIR filter for each sensor output so that the summation signal of the outputs of the FIR filters has zero for useless vibration modes. Some experiments were carried out, and good results were obtained.

Session 3SP

Speech Communication: Developmental Issues and Disordered Populations

Susan N. Nittrouer, Chair

Department of Special Education and Communication Disorders, University of Nebraska, Omaha, Nebraska 68182

Chair's Introduction—1:00

Contributed Papers

1:05

3SP1. Transition duration effects on place perception in hearing-impaired listeners. René A. E. Zakia and Arlene E. Carney (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

The purpose of this study was to determine the contribution of transition duration in signaling an alveolar/velar place-of-articulation contrast to listeners with mild-to-moderate sensorineural hearing loss. In the experiments reported here, *F1*, *F2*, and *F3* transitions were varied between values appropriate for [de] and [ge]. These were combined with transition durations ranging from 20 to 50 ms in 5-ms steps. In three separate identification tasks, listeners were presented with: (1) all levels of formant pattern varied orthogonally with all transition duration values; (2) all levels of formant pattern combined with an ambiguous duration value; and (3) all levels of transition duration combined with an ambiguous formant pattern. A trend was observed for hearing-impaired listeners to identify stimuli with shorter transition durations as [de] in the absence of other unequivocal cues for place of articulation. Overall performance of hearing-impaired listeners was more variable than that of normal listeners, predicted in part by the results of Summerfield *et al.* [Speech Commun. 4, 213–229 (1985)], in which hearing-impaired listeners showed a decreased ability to identify place of articulation from transition information alone. Results of these experiments will be discussed in terms of current views of temporal processing in hearing-impaired listeners. [Work supported by NIH.]

1:20

3SP2. Perception of synthesized vowel pairs by normal and specifically language impaired (SLI) children. Rachel E. Stark (Audiology and Speech Sci., Purdue Univ., West Lafayette, IN 47907) and John M. Heinz (J. F. Kennedy Inst., Baltimore, MD)

It has been proposed that SLI children have a rapid-rate auditory-processing deficit, and alternatively, that they have special difficulty with signals that are low in information or degraded. In the present study, steady-state vowel pairs, /a/-/i/ and /æ/-/ε/, were presented to 20 normal and 20 SLI children within a series of tasks, namely, discrimination, in which they had to detect change/no-change in a series; identification where they identified stimuli presented one at a time; and sequencing, where they indicated order of presentation of paired stimuli. For both vowel pairs, short (40–5 ms) and long (240–40 ms) series were presented. It was found that the performance of the SLI children was not significantly different from the normals' in tasks involving /a/-/i/. Both groups of children had greater difficulty with the /æ/-/ε/ than the /a/-/i/ tasks, but the SLI children showed significantly greater decrement in performance than normals for /æ/ vs /ε/. Neither group showed a significant effect of vowel duration. Implica-

tions of these findings for child language impairment will be discussed. [Work supported by NIH.]

1:35

3SP3. Effects of rate and bite-block manipulations on relationships between duration and variability in children's and adults' speech. Bruce L. Smith (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2299 Sheridan Rd., Evanston, IL 60208)

Based on the assumption that speakers are less skilled at talking under atypical conditions, segment duration and variability were studied in 5 and 11 yr olds and a group of adults as they produced short phrases in normal, fast-rate, and bite-block conditions. Preliminary analyses indicate that for all three groups, the bite-block task generally did not cause much difference in duration or variability from the normal condition. Although duration and variability both decreased in the fast-rate condition for all groups, relative changes in variability were typically less than those for duration. Several other comparisons of duration and variability also did not show very strong relationships between these two measures. For instance, variability was also evaluated as a function of intrinsic vowel duration. Although high vowels were approximately 30% shorter than low vowels for the three groups, variability was only about 15% less for high than for low vowels. For individual subjects, correlations between duration and variability were also generally rather low. Implications for speech production development will be discussed.

1:50

3SP4. The developmental role of spectral relations in processing place of articulation of nasal consonants from short duration segments. Ralph N. Ohde and Marleen T. Ochs (Div. of Hear. and Speech Sci., Box 552, Station 17, Vanderbilt Univ., School of Medicine, Nashville, TN 37232)

In processing place of articulation of prevocalic nasal consonants, the spectral change from the murmur to vowel onset is a critical cue for both child and adult speech. In order to further our understanding of the nature of this cue in the speech of children and adults, three children, ages 3, 5, and 7, and an adult female and male produced CV syllables consisting of either /m/ or /n/ in the context of four vowels /i,æ,u,o/. Ten adults identified the place of articulation of the nasal before and after the insertion of periods of silence between 25-ms segments of the murmur and vowel transition. The major findings were as follows: (1) across speakers, the average identification of isolated segments was low ranging from 70% to 53% for the murmur, and 71% to 49% for the vowel transition; (2) across speakers, the average identifi-

cation of the murmur + transition segment was above 90%; and (3) the insertion of temporal gaps caused a reduction in the identification of nasals with the greatest effects occurring for children, the alveolar place of articulation, and a silence gap of 150 ms. The findings will be discussed relative to differences between children and adults, and the potential mechanisms involved in processing spectral relations. [Work supported by NIH, DC00464.]

2:05

3SP5. The developmental precedence of spectral and temporal cues in the perception of the /r-w/ phonemic contrast. L. K. Fitzgerald and E. B. Slawinski (Dept. of Psychol., Univ. of Calgary, 2500 University Dr., N.W., Calgary, Alberta T2N 1N4, Canada)

The present study examined acoustical and developmental influences on the categorical perception of /r/ and /w/. One group of adults, and three groups of normally developing preschool children, aged 3, 4, and 5 years, respectively, participated in three experimental conditions. These conditions involved performing a video-game identification task employing three different synthetic "rock-walk" continua. In condition I, the acoustic tokens varied according to both spectral and temporal cues; in condition II, the acoustic tokens varied only with respect to spectral cues; in condition III, the acoustic tokens varied only with respect to temporal cues. As hypothesized, changes in perceptual responses were seen as a function of both age and acoustical cue availability. Temporal cues, relative to spectral cues appeared to be the more salient cue for 3 yr olds, whereas for the 4 yr olds, 5 yr olds, and adults the two types of cues appeared to be equally salient. Other developmental trends evidenced included findings that children, relative to adults, demonstrate greater /r/ perceptual categories, more variable perceptual

performance, and a tendency to base their phonemic categorizations on primarily only one cue.

2:20

3SP6. Age-related differences in processing dynamic acoustic cues to initial stops. Robert Allen Fox, Jeanne Gokcen, and Lida G. Wall (Div. of Speech and Hear. Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002)

The present study examines possible age-related differences in the ability to process rapidly changing acoustic information in the identification and discrimination of initial stop consonants. As described in Fox *et al.* [J. Acoust. Soc. Am. **89**, 1935(A) (1991)], older adults show lower accuracy rates in identifying both consonants and vowels in so-called "silent-center" tokens. Such results are compatible with the suggestion that older listeners have greater difficulty than younger listeners in processing dynamic acoustic cues, such as those represented by consonant transitions. Two sets of stimuli were created: one set represented a 7-step [be]-[de] continuum with a 40-ms transition followed by a 150-ms steady-state vowel; the second set was composed of a 7-step [be]-[de] continuum with the same transition followed by a 10-ms vowel. There were two sets of normal hearing listeners including 15 listeners aged 20-24 and 15 listeners aged 60-75. There was an identification task and two AX discrimination tasks (with either a 500- or 2000-ms ISI). Older listeners showed shallower slopes in their identification functions than did the younger listeners, particularly in the stimuli composed primarily of formant transitions. In the discrimination task, older listeners were significantly less sensitive to formant transition differences than were the younger listeners, but there were no significant differences related to interstimulus interval. [Work supported, in part, by an NIA grant to R. Fox.]

2:35-2:50

Break

2:50

3SP7. Vowel boundaries for steady-state and transient formants. Anna K. Nabelek, Zbigniew Czyzewski, and Hilary Crowley (Dept. of Audiology and Speech Pathology, Univ. of Tennessee, Knoxville, TN 37996-0740)

Fifteen-step /I-ε/ continua were generated to investigate effects of: (1) listening conditions (quiet, noise, and reverberation), (2) subjects' hearing (normal and impaired), and (3) trajectories of *F1* and *F2* (all steady-state, *F1* changing in upward or downward direction, and both *F1* and *F2* changing in upward or downward direction). Stimuli (each repeated 10 times) generated with the Klatt synthesizer were 200 ms long. Speech-spectrum noise was mixed at S/N = 0 dB. Reverberation was generated by a computer program and convolved with the stimuli (*T* = 0.9 s). Subjects, 10 in each group, were tested individually. Stimuli were delivered monaurally through an earphone at a comfortable level. Listening condition had no effect on the boundary location for either group of subjects and had no effect on boundary slope for the normal-hearing subjects but a significant effect on slope for hearing-impaired subjects. Type of trajectories had a significant effect on location of the boundary only for the hearing-impaired subjects and on boundary slope for both groups of subjects; the two continua with *F1* and with *F1* and *F2* changing in downward direction produced different results than the continua with steady-state formants and with formants in upward direction. [Work supported by NIDCD.]

3:05

3SP8. Intelligibility of speech sounds in amplitude-modulated noise. Theodore S. Bell, Laurie S. Eisenberg, and Donald D. Dirks (UCLA School of Medicine, 31-24 Rehabilitation Ctr., Los Angeles, CA 90024-1794)

Two experiments describe the effect of amplitude-modulated (AM) noise on consonant recognition in normal-hearing and hearing-impaired listeners. The paradigm made direct estimates of release from masking due to fluctuating noise by comparing performance from steady-state and AM conditions while controlling signal level differences. In the first experiment, ten young audiotically normal adults identified nonsense syllables varying in vowel context (a, i, u), order (CV, VC), and speaker (male, female) at -2 dB S/N in a shaped noise. The noise was either at a constant level or amplitude modulated by a sinusoid varying in frequency (10, 31.5, and 100 Hz) and depth of modulation (50% and 90%). Depth of modulation produced a significant improvement in intelligibility over steady-state performance, with the 90% modulation depth showing greater release than 50% modulation. The greatest improvement (14.9%) occurred with the 31.5-Hz modulation rate. Frequency effects varied significantly by speaker, and the interaction between frequency and depth of modulation was also significant. Depth of modulation interacted in complex ways with vowel, order, and speaker. Experiment 2 compared normal and hearing-impaired listeners with 90% AM noise at 31.5 Hz. The hearing-impaired individuals displayed

significant release from masking, although less than the control subjects. The amount of release varied by audiometric configuration. Overall, these results indicate that release from masking is evident in some hearing-impaired listeners and absent in others; audibility and temporal factors will be discussed. [Work supported by NIH.]

3:20

3SP9. Acoustic analyses of contrastive stress production in children with normal and impaired hearing. Arlene E. Carney (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131), Amy L. Weiss (Univ. of Iowa, Iowa City, IA 52242), and Richard Fahey (Boys Town Natl. Res. Hospital, Omaha, NE)

The purpose of this study was to determine the acoustic cues used by listeners making judgments of contrastive stress production in two groups of speakers: (1) children with normal hearing ranging in age from 3 to 6; and (2) children with severe-to-profound hearing loss ranging in age from 4 to 18 who were language matched to the normally hearing group. Children performed a picture-description task of pairs of photographs that varied in agent, action, or object. The second sentence of the pair was designed to have one semantic category as the stressed target. Perceptual judgments of stress in these speakers were reported in Weiss *et al.* [J. Speech Hear. Res. 28, 26-35 (1985)]. Subjects were divided into four groups according to hearing status and high or low listener agreement of produced stress. Measures of fundamental frequency (F_0), relative intensity, and relative duration were made. Both normally hearing and hearing-impaired groups had speakers who increased F_0 for stressed words within an utterance as well as those who did not vary F_0 or varied nontarget words. In addition, F_0 rise was observed less frequently in the object (word final) position for all speakers. Duration and intensity effects were more variable for both groups. [Work supported by NIH.]

3:35

3SP10. Acoustic parameters of nasal utterances in the hearing-impaired and normal-hearing speakers. Marilyn Chen (Res. Lab. of Electron. and Dept. of Elec. Eng. and Comput. Sci., MIT, Cambridge, MA 02139)

One of the more prevalent abnormalities in the speech of the hearing-impaired that contributes to reduced intelligibility is inadvertent nasality. From acoustic analysis of the speech of hearing-impaired children, the presence of an extra pole-zero pair between the first and second formants and reduced first formant prominence characterize spectra of nasalized vowels. The difference between the amplitude of the first formant and the amplitude of the extra peak, A1-P1, was a measure that correlated with listener judgments of the degree of vowel nasalization. To obtain further validation of these parameters as measures of nasalization, A1 and the pole-zero spacing were systematically manipulated in synthetic utterances. Perceptual experiments with synthesized, isolated static vowels [i], [I], [o], and [u] showed that both parameters contributed to the perception of nasality. Another perceptual experiment with a number of synthesized words (of the form bVt) gave results indicating somewhat different relative importance of the two parameters. Correlation of A1-P1 with the average nasality perception judgments of ten listeners was found for both groups of stimuli. [Work supported in part by NSF and NIH.]

3:50

3SP11. Articulatory compensation in hearing-impaired speakers. Melanie M. Campbell, Arthur Boothroyd (Dept. of Speech and Hear. Sci., City Univ. of New York Graduate Ctr., 33 West 42 St., New York, NY 10036), Nancy S. McGarr (Haskins Lab., New Haven, CT), and Katherine S. Harris (City Univ. of New York Graduate Ctr.)

Previous studies have shown that normal adults compensate for a

bite block at the onset of vowel production. It has thus been presumed that auditory feedback plays no role in on-line, speech-motor control. Some investigators have further concluded that while acoustic feedback may play some role, it is not a substantial component in the coordination of reciprocal articulatory movement. The role of auditory feedback is explored in this investigation of compensatory skills in oral subjects with congenital hearing loss. Six severely hearing-impaired and six profoundly hearing-impaired adults were recorded. The stimuli were repetitions of a carrier phrase containing a target word with one of three vowels (/i/, /I/, /ae/) in mixed, randomly selected sequences in four conditions: normal (with hearing aids), masking noise (without hearing aids), bite block (with hearing aids), bite block plus masking noise (without hearing aids). Compensation was measured by comparison of the first three vowel formants of each test condition with those of the normal condition. Formants were acoustically analyzed using LPC techniques. Preliminary data suggest that the first few tokens produced with bite block result in some changes in F_0 and loudness. However, there is not convincing evidence of failure of compensation. [This work was supported by NINCDS Grant No. DC00121-29.]

4:05

3SP12. Effects of distorted auditory feedback on speech. Vivien C. Tartter, Nassima Abdelli, and Alexandra Economou (Dept. of Psychol., City College, 138 St. at Convent Ave., New York, NY 10031)

A research program for acoustically measuring changes in speech production will be described. Here, F_0 and spectrographic measurements target acoustic correlates of consonant, vowel, and syllabic stress features in 45 phonemically balanced words. To date, 5 Nucleus-22 multichannel cochlear implant patients have participated (at least) prestimulation, immediately after stimulation, 3-6 months after stimulation, and 1-year after stimulation. Commonalities among patients include: a rapid increase and stabilization of F_0 (in females) to normal values; immediate and continuing changes in vowel and syllable durations and in relative amounts of breathy noise and low-frequency energy; gradual changes in vowel formant frequencies with movement still observed at 1 year; and changes in consonant properties (such as F_1 cutback for voiceless stops) emerging for the first time at 1 year. Results relate to (1) the potential role of critical periods in the effect of auditory feedback on speech habits, (2) the relative susceptibility of speech features to modification, and (3) the likely auditory parameters available from Nucleus-22 WSP III stimulation. [Work supported by NIDCD.]

4:20

3SP13. Acoustic characteristics of normal and dysarthric speech. Marios Fourakis and Carl D. Coelho (Commun. Disord., Gaylord Hospital, P.O. Box 400, Wallingford, CT 06942)

The experiment reported here examined the implementation of final lengthening and main sentence stress in the speech of one moderately ataxic dysarthric (AD) and one normal speaker. A set of C1VC2 syllables was constructed such that C1 varied over [b,d,g,p,t,k], C2 was always [d], and V was one of the nine nonretroflex vowels for the [b__d] context, and one of [i, e, æ] for the other five initial contexts. These syllables were embedded in carrier sentences such that the target syllable occurred in two positions, final and medial, and bearing main sentence stress or not. Eight repetitions of each syllable in each position-stress condition were recorded and digitized. Measurements of initial closure, VOT (if any), vowel, and final closure durations were made directly from the waveform. Preliminary results indicate that the AD speaker implemented final lengthening of the target syllable in a manner similar to the normal speaker overall, but differed in the distribution of the lengthening effect over the acoustic segments comprising the syllable. This result is not in agreement with the results reported by Bell-Berti *et al.* [Proc. of the 12th Int. Congr. Phon. Sci. (1991)] for French AD speakers. Results on the implementation of sentence stress will also be presented. These will be discussed within the framework of how cerebellar deterioration can affect the fine control of speech timing.

3SP14. Articulatory and acoustic study of consonant production in a traumatically brain injured speaker. Fredericka Bell-Berti (St. John's Univ., Jamaica, NY 11439 and Haskins Lab., New Haven, CT 06511), Maria Rossetti (Haskins Lab., New Haven, CT 06511 and CUNY Graduate School, New York, NY 10036), Kiyoshi Oshima (Haskins Lab., New Haven, CT 06511), and Carl Coelho (Commun. Disord. Dept., Gaylord Hospital, Wallingford, CT 06492)

This study examined the clinical utility of instrumental studies of the speech of neurologically impaired persons, as well as documented the particular articulatory patterns of one individual, a right-handed 36-yr-old female, 5-yr post onset of TBI, with right hemiparesis. Communicative impairments included relatively mild cognitive deficits and moderate-to-severe hypokinetic dysarthria. The speech sample included repetitions of isolated sustained continuant consonants, of VCV disyllables, and of sentences containing consonant clusters. The articulatory data were electropalatographic (EPG) signals recorded using a RION soft artificial palate with a sampling rate of 64 fps. The acoustic signals were recorded simultaneously with the EPG signals and subjected to spectral analysis. Vowel environment was found to have a strong influence on lingual contact patterns for /s/. In general, within vowel height groups, the extent of lingual contact for /s/ was greater in the VCV context than in words produced in sentences. Spectral patterns associated with differences in lingual contact for /s/ will be described. [Work supported by NIH Grant No. DC-00121 to Haskins Lab.]

3SP15. Resolution of part-word repetitions by stutterers: A temporal-spectral study. N. S. Viswanath and Amy Neel Beardsley (Stuttering Ctr., Dept. of Neurology, Baylor College of Medicine, Houston, TX 77030-3498)

Forty stuttering events on words beginning with /k/ and 17 with /g/ obtained from five adult male stutterers were examined from the pause before the first fragment to the eventual fluent production of the word. Spectrographic measurements made using CSL (Kay) included: duration of stuttering events, duration of pauses in stuttering events, duration of each fragment and its component parts if any (frication, aspiration, vowel), spectra of the release burst, and spectra of the following vowel at three predefined points on LPC signal. The findings were: 83% of events were composed of one or two fragments, fragments in /k/ plosive environments less frequently had vowels following release (18%) than fragments in /g/ environments (50%), pauses were short (77% under 200 ms) and may reflect articulatory closure duration rather than linguistic processes, and peak values of burst spectra did not progressively approximate those of fluent productions in two-fragment stuttering events. These and other results highlight the variable nature of articulatory processes in the resolution of part-word repetitions and provide constraints for modeling of stuttering events. [Work supported by a USPHS grant.]

TUESDAY AFTERNOON, 12 MAY 1992

SALON D, 12:55 TO 4:45 P.M.

Session 3UW

Underwater Acoustics: Matched-Field Processing

John M. Ozard, Chair

Defense Research Establishment Pacific, FMO Victoria, British Columbia V0S 1B0, Canada

Chair's Introduction—12:55

Contributed Papers

1:00

3UW1. Performance stability of high-resolution matched-field processors to sound-speed mismatch in a shallow-water environment. G. B. Smith, H. A. Chandler, and C. Feuillade (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

The effects of variations in water sound-speed parameters on the performance of four matched-field processing algorithms (Bartlett, maximum likelihood, sector focusing, and multiple constraint) have been investigated. The SNAP propagation model was used to generate the replica acoustic pressure field for a shallow-water channel with a depth variable sound-speed profile typical of a midlatitude summer environment. It was also used to simulate a "detected" field due to an acoustic source. These were then correlated using the four algorithms for selected degrees of mismatch of the water sound-speed profile. The maximum likelihood and multiple constraint estimators achieved good peak resolution and high accuracy for small degrees of mismatch. The maximum likelihood estimator deteriorated quickly as the mismatch increased. The multiple constraint performed significantly better, but

both failed to give correct location estimates at the higher mismatch values. In contrast, the sector focusing estimator continued to give accurate location estimates over the whole range of sound-speed mismatch, with slightly less peak resolution at small mismatch values. It also performed much better overall than the Bartlett estimator, which was found to give accurate location estimates over the whole range of mismatch values used, but with poor peak resolution against a background containing many high sidelobes. [This work was supported by the Naval Res. Lab.]

1:15

3UW2. Effects of sound-speed mismatch in the lower water column on matched-field processing. Ellen Livingston (Naval Res. Lab., Washington, DC 20375-5000)

The sensitivity of matched-field processing to sound-speed mismatch in the upper water column (above 1000 m) has frequently been discussed [A. Tolstoy, *J. Acoust. Soc. Am.* **85**, 2394-2404 (1989); Henrik

Schmidt *et al.*, J. Acoust. Soc. Am. 88, 1851-1862 (1990)]. In this work, the effects of sound-speed mismatch in the lower water column (3000-5000 m) are investigated. Such mismatch can arise from differences in climatology data or inaccuracies in the sound-speed equation. Low-frequency sources are simulated at shot depths in a range-independent deep-water Pacific environment and are processed with a mismatched replica field on a long vertical receiving array. Source range is varied from 50 to 500 km. For a sound-speed mismatch of 0.6 m/s at 5000 m, the effects on the mainlobe peak level are strongly and cyclically range dependent, varying by as much as 1 dB. A closed form analysis of this phenomenon is presented and the effects of source depth and of array placement in the water column is considered.

1:30

3UW3. Limitation of reflecting irregular bathymetry on matched-field processing. O. Diachok and J. F. Smith (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. At very low frequencies these modes are significant over: thickly sedimented bottoms and thinly sedimented bottoms where $V_s > V_w$. In thinly sedimented regions of the mid-pacific, V_s is approximately 1570 m/s (Diachok, 1991). Results of a study to determine the effect of experimentally obtained bathymetry in the thinly sedimented Pacific on matched-field processing follow. The bathymetry exhibits an irregular variation in depth as a function of range and is assumed to have a constant sound speed. The range- and depth-dependent sound speed in the water column was obtained from tomography data. 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 (0.15 dB at 1000 km at 15 Hz) in matched-field studies against "exact" results produced by the normal modes program KRAKENC in a range-independent environment. Simulation of the loss in signal gain was conducted at a frequency of 14 Hz at various ranges and constant bottom sound speeds of 1530, 1540, 1550, 1560, and 1570 m/s. The effects of irregular bathymetry and shifts in critical angle were to increase signal array degradation and reduce the accuracy of 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.

1:45

3UW4. Holographic array processing in a range-dependent ocean. Azmi A. Al-Kurd, Robert P. Porter, and Pierre D. Mourad (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Imperfect knowledge of the salient characteristics of the propagation medium limits the performance of acoustic array processors at long ranges in the ocean. Holographic and phase conjugation techniques can be used to diminish the range-integrated effect of the medium and reconstruct the wave front in the vicinity of a scatterer or other signal source. Then, using a back-propagation technique that focuses sound at the position of the unknown source, the location of the source can be determined. In this paper, the WKB approximation is implemented for a range-dependent ocean with both N^2 bilinear and exponential stratification of the sound-speed profile. The results of the analysis and simulation show the possibility of localizing a source at large distance with great accuracy.

2:00

3UW5. Array processing with acoustic measurements at a single point in the ocean. G. L. D'Spain and W. S. Hodgkiss (Marine Phys. Lab., Scripps Inst. of Oceanogr., San Diego, CA 92152-6400)

In order to obtain information on the spatial dependence of the underwater acoustic pressure field, the traditional approach has been to deploy a spatially distributed set of hydrophones. However, by use of a Taylor series expansion, the spatial dependence of the sound field can also be obtained from measurements at a single point in space. The purpose of this presentation is to discuss the adaptation of high-resolution array processing algorithms to such single-point measurements. Simulation results will be presented to illustrate the application of Capon's minimum variance method, the linear prediction method, and the multiple signal classification scheme to measurements in a Taylor series expansion to first order and in an expansion to second order of the pressure field. In addition, a comparison will be made of the high-resolution methods to standard processing (equivalent to the Bartlett method) using actual ocean acoustic data from geophones and hydrophones. Extensions of these techniques to data collected by a receiving system that combines the use of spatially distributed elements with the Taylor series expansion approach will be discussed. [Work supported by ONR and ONT.]

2:15

3UW6. Matched-field inversion of geoacoustic parameters of elastic bottoms. John W. Wolf and Orest I. Diachok (Naval Res. Lab., Washington, DC 20375)

The potential of matched-field inversion of ocean bottom parameters is demonstrated through simulation of signals transmitted by one ship and received on a 1.5-km-long horizontal ship-towed array in end-fire mode. The simulations use an ocean model that is representative of a thinly sedimented (36-m) basalt bottom. The output of a matched-field processor (Capon correlator) at a known source position is maximized by searching through bottom parameter differences from a geoacoustic profile. In particular, the shear velocities, C_s , in a model of the upper crust and an overlying thin sediment layer are varied about the true values. Relatively small differences (15% of true) in basement shear velocity produce a 4-dB degradation in processor output. Shear speed mismatch in the sediment produces thin film effects that appear in the processor output function as resonance peaks at several values of ΔC_s . The values of ΔC_s for most peaks vary strongly with source frequency except the peak at $\Delta C_s = 0$. The results illustrate the potential of broadband matched-field processing to invert elastic bottom parameters, particularly shear speeds in thin sediments and upper crust, which to date have generally proven inaccessible with existing methods.

2:30

3UW7. Matched-field inversion of geoacoustic model parameters using unguided simulated annealing. Colin Lindsay and N. R. Chapman (Defence Res. Establishment Pacific, FMO Victoria, BC V0S 1B0, Canada)

A method has been developed for the estimation of geoacoustic model parameters by inversion of acoustic field data using a nonlinear optimization procedure based on simulated annealing. The cost function used by the algorithm is the Bartlett matched-field processor which relates the measured acoustic field with replica fields calculated by the SAFARI program. Model parameters are perturbed randomly over a specified range, and the algorithm searches the multidimensional parameter space of ocean bottom models to determine the parameter set corresponding to the best replica field. Convergence is driven by adaptively restricting the search space to regions with above-average values of the matched-field correlation. This approach removes the need for a predetermined annealing schedule. The performance of the method is demonstrated for a vertical line array in a shallow-water environment where the bottom consists of homogeneous elastic solid layers. Simulated data were used to study the effects of noise contamination, under-sampling of the pressure field and uncertainty of the experimental geometry on parameter estimation. Results are presented for the inversion of data obtained in an experiment off the West Coast of Vancouver Island.

3UW8. Replica partitioning and preselection for faster matched-field processing. John M. Ozard, Pierre Zakarauskas, Don G. Berryman (Defence Res. Establishment Pacific, FMO Victoria, BC V0S 1B0, Canada), and Michael J. Wilmut (Royal Road Military College, FMO Victoria, BC V0S 1B0, Canada)

Matched-field processing requires the evaluation of the inner product of the measured and replica fields at a grid of points covering the search region. This process may require a significant computational time despite the increased speed of computers. A way of reducing the

search space is described and evaluated. First, a basis for the signal space is found for all possible source positions. Then the replica fields for the search region are partitioned into groups with similar excitations of the basis vectors used to represent the replicas. One can then quickly select the group of replicas that is most like any data to be matched, and then do a detailed search over only that group to find the best match. An example is shown of the detection and localization performance of the Bartlett beamformer with and without preselection of the replicas. A reduction in computation time of 10 to 20 was achieved with only a slight loss in performance over a wide range of signal-to-noise ratios.

3:00–3:15

Break

3:15

3UW9. Matched-field processing speedup through fast nearest-neighbor search in excitation space. Pierre Zakarauskas, John M. Ozard, Don G. Berryman (Defence Res. Establishment Pacific, FMO Victoria, BC V0S 1B0, Canada), and Mike Wilmut (Royal Road Military College, Victoria, BC V0S 1B0, Canada)

By casting the matching portion of the MFP problem into a nearest-neighbor search, one may apply the fast nearest-neighbor search algorithm to obtain potentially considerable speedups. This algorithm makes it possible to find the replica closest to a test pattern (peak of the Bartlett processor output), in a time which is asymptotically constant with the number of replicas. The algorithm first partitions the excitation space, and finds which partition each of the replicas falls into. When a test pattern is supplied, one compares it to all replicas within the partitions which are within a given distance R to the test pattern. If the distance d between the closest replica found during the first pass and the test pattern is less than R , then the search stops. If not, then a second pass is done using d as a search parameter to select partitions. This algorithm guarantees that the nearest neighbor is found. The mean number of operations done to find the nearest neighbor is presented as a function of the density of patterns per partition and the dimensionality of the excitation space. A global minimum is shown to exist for the mean number of operations at a partition size corresponding to slightly less than one replica per partition.

that is based on D. Chu's exact solution for a wedge [J. Acoust. Soc. Am. **86**, 1883–1896 (1989)]. S. Li and C. S. Clay have also given a discussion of single channel theory and experiment [J. Acoust. Soc. Am. **82**, 1409–1417 (1987)]. The results show that source localization in a wedge is significantly promoted by the range dependency of the environment, which eliminates "range" sidelobes, even for a single hydrophone. The signal peak is more effectively resolved by locating the receivers as far from the wedge apex as possible. "Angle" sidelobes may be effectively eliminated by introducing additional receivers in an arc at constant range from the apex. [This work was supported by the Naval Res. Lab. and by the Office of Naval Res. under Contract No. N00014-89-J-1515 (CSC).]

3:45

3UW11. A matched-mode processing technique for localizing a transient source. I-Tai Lu and Han-Yang Chen (Dept. of Elec. Eng., Weber Res. Inst., Polytechnic Univ., Rte. 110, Farmingdale, NY 11735)

A matched-mode processing method separates the range and depth estimation tasks, thereby obtaining a sequential procedure that circumvents the need for a two-dimensional search over all hypothesized source locations. Therefore, when applicable, it is more efficient than a matched-field method. Here, the conventional frequency-domain, matched-mode method is extended to deal with more realistic nonharmonic, narrow-band signals directly in the time domain. The waveform of each mode is derived by a least-squares estimating procedure in the time domain. This procedure is essentially equivalent to the conventional modal filtering approach which is carried out by inner products of the received signals and the modal function in the frequency domain. Then, the source range and depth information can be estimated by the arrival times and magnitudes of some *clean* modal arrivals. The advantage of this extension is that the arrival times of the modal wave packets are less sensitive to model uncertainties than the phase of the different modes in the purely harmonic case. Moreover, this approach does not require a prior knowledge of the source waveform. This extension is applicable when the source signal is narrow band and the range dependence in the ocean environment is weak. In numerical simulation, the source and receiver arrangement and the ocean parameters are chosen from the experimental data measured in the Mediterranean Sea [S. M. Jesus, J. Acoust. Soc. Am. **90**, 2034–2041 (1991)]. [Work supported by ONR.]

3:30

3UW10. Acoustic source localization in a wedge waveguide using time-domain techniques. C. Feuillade (Naval Res. Lab., Stennis Space Center, MS 39529-5004) and C. S. Clay (Geophysical and Polar Res. Ctr., Univ. of Wisconsin–Madison, Madison, WI 53706)

In previous work [C. Feuillade and C. S. Clay, J. Acoust. Soc. Am. Suppl. **1** **88**, S26 (1990)], the results of an investigation into time domain source imaging techniques in a range-independent, shallow-water waveguide were reported. The research reported here extends this method to consider the case of an impulsive source in a density penetrable wedge waveguide. Single-channel source location information is used by means of time domain transmissions to multiple receivers. The signals from the receivers are time reversed and transmitted through sources at the receiver locations. The field is mapped to locate the source. Particular attention is given to the peak amplitudes, sidelobe amplitudes, and spatial resolution as a function of receiver channels. The field calculations are performed by using an image approximation

4:00

3UW12. A back-propagating ray technique for source localization. I-Tai Lu and Peter Voltz (Dept. of Elec. Eng., Weber Res. Inst., Polytechnic Univ., Rte. 110, Farmingdale, NY 11735)

This paper reports on a novel source localization scheme in the time domain consisting of the following two steps: (1) use advanced signal processing techniques to sort out the ray arrivals, including their arrival direction and arrival times; and (2) back propagate these rays to their source region by a ray tracing algorithm. Some key advantages of this approach are: (1) Localization ambiguity is minimized by utilizing ray travel times as well as path convergence. (2) Ray tracing is a very efficient propagator in range-dependent environments. (3) The approach does not require full-wave matched-field processing and the required knowledge about the environment is minimized. (4) Phase ambiguity is eliminated by employing the group delay of arriving wave packets. (5) Ray chaos can be avoided simply by excluding chaotic rays from the backpropagating procedure. [Work supported by ONR.]

4:15

3UW13. Active matched-field processing. J. S. Perkins, W. A. Kuperman, L. E. Tinker (Naval Res. Lab., Code 5160, Washington, DC 20375-5000), and G. T. Murphy (Planning Systems Inc., McLean, VA 22102)

In *passive* matched-field processing, a model is used to simulate the acoustic field propagating from an object to a receiver array (usually with vertical extent). After generating many replica fields corresponding to many object positions, one can estimate the location of the object by seeing which of these replica fields best matches the received data. In this paper, *active* matched-field processing is demonstrated. An acoustic model based on adiabatic normal modes is used to simulate the acoustic

field on a vertical receiving array in an active scenario. That is, the acoustic energy from a point source propagates to an object in a three-dimensional environment, is scattered by the object, and is subsequently received on a vertical array of hydrophones. Just as in the passive case, this model is used to create many replica fields corresponding to many object positions, and then the position of the object is estimated by determining which of these replica fields best matches the received data. In principle, it is also possible to determine the orientation of the object.

4:30

3UW14. Matched-field localization of acoustic backscatter in the Arctic Ocean. Thomas J. Hayward (Naval Res. Lab., Washington, DC 20375-5000)

Matched-field processing has been studied extensively in recent years as a method for localizing underwater sound sources. For a source in the water, matched-field processing is frequently implemented assuming a point source for the replica field, ignoring the directional radiation pattern of the source. Generalization of the method to boundary backscatter from ocean surface features requires computation of the scattered field of an extended-body scatterer; in general, one expects that the directional properties of the scattered field may need to be included in the replica generation. In this work, localization of surface (ice) backscatter using data collected during the CEAREX 89 Arctic field experiment is demonstrated, using first a point-source replica. Then, the possibility of improving resolution and sidelobe levels is investigated by using a replica for the scattered field generated by using methods of coupled normal mode theory. Performance improvement is first investigated in simulations, one purpose of which is to test the sensitivity of the method to the scatterer shape. The improved field replicas are then applied in matched-field processing using the real data. [Work supported by the ONR Arctic Sciences Program.]

Session 4AA

Architectural Acoustics: General Topics

Rollin O. Boe, Chair

2404 Washington Boulevard, Suite 800, P.O. Box 389, Ogden, Utah 84402-0389

Invited Paper

8:30

4AA1. A tribute to Richard N. Hamme: The father of open office acoustics. David A. Harris (Canoga Park, CA 91303)

The late Richard (Dick) Hamme, proprietor and founder of Geiger and Hamme Laboratories in Ann Arbor, Michigan was a Fellow of the Acoustical Society of America, and a leader and pioneer in the development of procedures for the evaluation of acoustical materials and architectural systems. Dick and his associates corrected major misconceptions about open plan office acoustics. Many said that achieving speech privacy in the open plan was impossible. Not only did his efforts prove entirely the opposite, he invented, developed, and nurtured a whole technology that is just now being understood by the acoustical community. Procedures developed by Hamme for the U.S. General Services Administration, Public Building Services were used by public and private sector specifiers alike. Known as PBS C.1, C.2, and C.3 with several ancillary standards, these documents led to the re-thinking of the acoustical specifications in terms of system performance. After many years of use, these documents have been recently adopted by ASTM Committee E-33, Environmental Acoustics. This paper describes the evolution of the achievement of confidential speech privacy, a process that began in the late 1960's and has been refined to a technology of its own.

Contributed Papers

9:00

4AA2. Sabine reverberation equation revisited. Robert W. Young (1696 Los Altos Rd., San Diego, CA 92109)

About 1900 W. C. Sabine reported the "sound absorbing power" of materials in a reverberant room as proportional to a logarithm of the ratio of an initial to a final "residual sound," after the source is stopped. In present-day terms, sound pressure level (a logarithm) decays at the rate d decibels per second, after the source is stopped. At least as far back as 1965, ASTM C423, "Sound Absorption... Reverberation Room Method," obtained the Sabine absorption A in metric sabins by $A = 0.9210 Vd/c$; $0.4 \ln 10 = 0.9210$ is a pure number. The volume of the reverberation room is V , cubic meters; the speed of sound of air therein is c , meters per second. On substituting reverberation time $T = 60 \text{ dB}/d$, Sabine sound absorption $A = 55.3(\text{dB})V/cT$; this dimensionally complete equation shows the decibel explicitly. The Sabine absorption of a surface of area S in square meters, and Sabine absorption coefficient a , is $A = Sa$. The unit of the Sabine coefficient a is thus the decibel; it is dimensionless in the sense of length, mass, or time; the unit is usually omitted. Being derived from a decay rate, the Sabine sound absorption coefficient is not "Ideally the fraction of the randomly incident sound power absorbed..."

9:15

4AA3. On the presentation of large amounts of architectural acoustics data. Richard H. Campbell (Worcester Polytech. Inst., Acoust. Lab., 100 Institute Rd., Worcester, MA 01609)

With the proliferation of data acquisition systems that process time

domain impulse responses taken in auditoria, there is an overwhelming quantity of (useful) data that can be computed using the digital signal processing algorithms ordinarily supplied with such systems. The example given is a 507-seat lecture hall where two different speech-reinforcement systems were evaluated at 10 spatial locations using MLSSA, generating over 4.8 megabytes of time domain data in less than 2 h. One additional hour of semiautomated postprocessing resulted in 960 important acoustic parameters from a field of 2100 pieces of analytical data, and 1960 modulation transfer function parameters for STI calculations. These multitudinous data are organized into a series of colorful graphs that show a large amount of interconnected information in a single view, and in a way that provides maximum insight to the experienced acoustician, yet can be created easily in a laboratory having a modest software budget and rather ordinary color display and printing means.

9:30

4AA4. Acoustic scale modeling of auditoria. Whitney N. Potter and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

Acoustic scale modeling was explored using a 1/48th scale model of the Lang Performing Arts Center at Swarthmore College [M. H. Soon and E. C. Everbach, *J. Acoust. Soc. Am.* **89**, 1898 (1991)]. A system was designed to test the scaled acoustic properties of modeling materials, and appropriate materials were selected to model to acoustic treatments of the auditorium. Measurements of acoustic descriptors made from the model were compared to measurements made in the actual auditorium. This modeling technique can be applied to testing designs

of unbuilt auditoria, or to testing possible acoustic treatments in existing auditoria.

9:45

4AA5. Specific property of sound field and computer simulation in a hemianechoic room. Biao Cai (Jiaotong Univ. of East China, Nanchang 330013, People's Republic of China) and Jiqing Wang (Inst. of Acoustics, Tongji Univ., Shanghai 200092, People's Republic of China)

This paper is based on the calculative method used in a former paper

[J. Wang and B. Cai, *J. Acoust. Soc. Am.* **85**, 1206–1212 (1989)]. With the help of a computer, the calculations of the sound field of a point source in a hemianechoic room show that the results are related to the height of the point source from the ground. Uncertainty of the interference field of direct sound and the first reflected sounds of pure tone is often greater than the requirements in ISO 3745. Authors computed the sound power level of the machine and the directivity factor of the point source (suppose the sound source has an omnidirectivity) with computer simulation. The systematic error of the sound field is greater than the requirements in ISO 3745. The results agree with the experiments. In accordance with simulated results, several kinds of ISO 3745 testing methods of the machine's sound power level were discussed.

10:00–10:15

Break

10:15

4AA6. The directional scattering factor: Experimental determination. Peter D'Antonio (RPG Difusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20772)

This paper suggests experimental procedures to determine a three-dimensional directional scattering factor (DSF) or directivity balloon, which fully represents the backscattered energy from any acoustical surface of arbitrary size, as a function of the frequency and direction cosine of the incident sound. The DSF is necessary in acoustical design, computer room modeling programs, and auralizations. D'Antonio [P. D'Antonio and J. Konnert, 79th AES Convention, Preprint No. 2295 (October 1985)] introduced a boundary measurement technique for evaluating directional scattering from absorptive, reflective, and diffusive surfaces using a pressure-zone microphone and the TEF analyzer. A more comprehensive automated polar mapping technique is suggested. This approach is capable of yielding the full 3-D backscattered directivity balloon using either the impulse response, transfer function, or intensity. A test sample approximately 20 m² in size is placed on the floor boundary, a hemispherical or flat grid of appropriately spaced microphones is placed above this at approximately 5 m, and the test signal sound source is suspended 5 m above the microphone grid. The microphone grid records the backscattered energy at an appropriate angular increment for each angle of incidence. These data are then analyzed and corrected for the effects of finite sample area, speaker and microphone responses, and near-field/far-field differences.

10:30

4AA7. A first experimental comparison between the Schroeder's backintegration method and a new method based on sound intensity for measuring reverberation time. Domenico Stanzial and Eleonora Carletti (Cemoter, Natl. Res. Council of Italy, Via Canal Bianco 28, I-44044 Cassana, Ferrara, Italy)

A new method based on the experimental evaluation of the energy transfer function of an enclosed space from sound intensity measurements has been experimentally compared with the well-known method of measuring reverberation time due to Schroeder [M. R. Schroeder, *J. Acoust. Soc. Am.* **37**, 409–412 (1965)]. The measurements of active and reactive sound intensity were carried out by a B&K 2133 real time frequency analyzer and the reverberation time was computed using the built-in reverbBack user defined function over a pressure input multi-spectrum. As the transfer function is simply related to the reverberation

time through an Eyring-like formula [D. Stanzial, *J. Acoust. Soc. Am. Suppl. 1* **88**, S185 (1990)] the comparison gives a first experimental confirmation of the intensimetric approach to reverberation time evaluation.

10:45

4AA8. Consulting applications for an acoustical simulation room. Richard G. Pierson (Orfield Associates, Inc., 2709 East 25th St., Minneapolis, MN 55406)

An acoustical simulation room was constructed for acoustical consulting use. The design criteria called for a space that could simulate the background noise, reverberation, spectral, time based, and directional response of proposed and existing spaces. Readily available professional audio and acoustical products were utilized for implementation into a dedicated space. These include programmable delays, reverberation processors, equalizers, and mixers as well as numerous amplifiers and loudspeakers. Significant use of digital audio interfacing was utilized to maximize system signal-to-noise ratios. Applications have included simulation of the acoustical response to architectural spaces and mechanical system noise for subjective evaluation; simulation of diffuse sound fields for attenuation evaluations; and simulation of a sound reinforcement system and masking response for intelligibility studies and subjective evaluation. Binaurally recorded demonstrations of the Orfield Associates Acoustical Simulation Room will be available at the end of the session.

11:00

4AA9. Adaptation to transformed auditory localization cues in a hybrid real/virtual environment. Barbara Shinn-Cunningham, Nat Durlach, and Richard Held (Res. Lab. of Electronics, MIT, 36-757, 77 Massachusetts Ave., Cambridge, MA 02139)

The ability of humans to adapt to intermodal discrepancies is an important factor in the design of virtual environments. In the present study, azimuthal localization cues were altered (to magnify azimuthal differences near zero degrees) relative to real proprioceptive, visual, and vestibular cues. Subjects were alternately tested and trained in hybrid real/virtual environments where auditory stimuli were synthesized (using a PC, Convolvotron, and electromagnetic head tracker) to be at one of 13 discrete positions marked by real lights. Testing consisted of identifying the azimuth of virtual sound sources without relevant visual

feedback or significant head motion. During training, a virtual sound from one of the positions was gated on with the light at that position, and subjects turned their heads to face the composite stimulus. Further identification experiments involving modified experimental procedures,

as well as minimum-audible-angle discrimination experiments, were performed to help clarify the results. Preliminary findings on resolution, bias, and identification will be briefly overviewed. [Work supported by AFOSR, Grant 90-200.]

WEDNESDAY MORNING, 13 MAY 1992

SALON J, 8:00 A.M. TO 12:00 NOON

Session 4EA

Engineering Acoustics: New Developments in Electroacoustical Instrumentation, Systems, and Applications

Victor Nedzelnitsky, Chair

National Institute for Standards and Technology, Sound Building 233, Room A147, Gaithersburg, Maryland 20899-0001

Chair's Introduction—8:00

Invited Papers

8:05

4EA1. High-speed digital, fiber-optic-linked, telemetry considerations for underwater acoustic arrays. David A. Desrosiers (Naval Surface Warfare Ctr., Dept. U42, 10901 New Hampshire Ave., Silver Spring, MD 20903-5000)

The effective design of an underwater cable-to-shore based telemetry system for large hydrophone arrays centers on efficient use of the following: the bandwidth of the transmission media, the information coding scheme, and the power at the remote end. Previous systems, using submarine coax cables 6 mi long, were bandwidth limited by cable attenuation to a 4 Mb/s transmission rate. This rate supported 20 data channels at a 5-kHz bandwidth (BW) with 8-bit resolution (48-dB dynamic range). The need for 100 channels in deeper water 20 mi offshore initiated the change to fiber-optic cable. The greater bandwidth of the fiber allows a transmission rate above 100 Mb/s, with 40 Mb/s actually being used. Dynamic range at 5 kHz BW was increased to 12-bit resolution (72 dB). Utilizing advancements in communications, data acquisition and low power electronics permitted a modular, bus-structured system, designed with off-the-shelf electronics. These improvements were first implemented in March 1991 in 1200-ft-deep water on the end of a 20-mi-long fiber-optic submarine cable, with a similar system still in operation today.

8:25

4EA2. A precision measurement system for use in underwater acoustics measurements. S. E. Forsythe and G. L. Hansen (Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

A Precision Measurement System (PMS) has been developed at NRL-USRD to serve both as a production measurement system and as a platform for ongoing research in new signal generation and measurement techniques. This system is intended as a single-platform solution to most of USRD's underwater acoustic test needs. Calibration of transducers and measurements of acoustic coatings are included in the tests performed by the PMS. The precision measurement system offers the following features: wide range in both frequency and amplitude; built-in calibration for traceability to NIST; multiple channels for simultaneous signal measurements; a flexible combination of analog gain and filtering stages; digital technology to allow generation of complex waveforms as well as sophisticated processing of measured data; an integrated programming language interface based on VAX/VMS workstations; a variety of signal processing algorithms for wideband versus narrow band, continuous versus transient, parametric versus nonparametric signal processing techniques. Four copies of the system have been built to insure uniformity of measurements across all USRD measurement facilities. Because the system is based on VAX/VMS, a variety of software is available for plotting and reporting as well as more sophisticated offline mathematical analysis.

8:45

4EA3. New standards at NIST based on digital techniques. Barry A. Bell (Electricity Div., Natl. Inst. of Standards and Technol., Dept. of Commerce, Technol. Admin., Gaithersburg, MD 20899)

The ability to accurately characterize the linearity, effective bits, step response, and other properties of analog-to-digital converters (ADCs), waveform recorders, and high-speed digitizers is critical in many measurement applications, including some electroacoustical ones. Test methods and techniques have been developed at NIST for evaluating the performance of high-resolution (up to 18 bits), high-accuracy (to 2 ppm) ADCs and digital-to-analog converters (DACs), as well as 4- to 12-bit high-speed (up to 500 Msamples/s) ADCs, typically used in sampling oscilloscopes and waveform recorders. Several of these NIST-developed methods have been incorporated in the recent IEEE Std. 1057 Trial Use Standard for Digitizing Waveform Recorders (July 1989). Precision DACs having small linearity errors can be used to provide reference electrical waveforms. By using a pair of such DACs, a 2-Hz to 50-kHz audio frequency phase angle calibration standard was developed at NIST that demonstrated the feasibility of generating dual sinusoidal waveforms with a relative phase difference resolution of about 2-mdeg and an accuracy of better than 10 mdeg. With careful attention to the fidelity of each step in the waveform, a dual-DAC approach for interleaving the small steps of the output waveform has formed the basis for an rms ac voltage standard with a frequency range from less than 1 Hz to 50 kHz, good to an overall rms uncertainty of 20 ppm.

9:05

4EA4. Application of dynamic signal analyzers to selected electroacoustical measurements. Victor Nedzelnitsky (Natl. Inst. of Standards and Technol., Sound Bldg. (233), Rm. A147, Gaithersburg, MD 20899-0001)

Modern dynamic signal analyzers incorporating a variety of digital signal processing techniques and excitation source types are now sufficiently accurate to be used in many precision measurement situations for which much more labor-intensive methods previously have been necessary. Such use typically requires well-designed analyzers for which individual electrical components and system performance have been well characterized, and may require performing the same type of measurement several times with different choices of analyzer setup parameters, involving different tradeoffs among measurement uncertainty components. Validation of analyzer measurement results by comparison with results of established techniques is often necessary. These analyzers also can provide new ways to examine the individual influences of some measurement uncertainty components that have been notoriously difficult to quantify. Such components include those attributable to imperfections in anechoic chambers, and to unwanted scattering and diffraction effects of instrument structures. Specific examples to be considered include preliminary measurements of free-field phase characteristics at low frequencies in the large NIST anechoic chamber, and the effects of chamber imperfections and of slight standing waves between opposing microphone diaphragms during free-field calibration by reciprocity at essentially normal incidence in the small NIST anechoic chamber at frequencies up to 100 kHz.

9:25

4EA5. Measuring hearing-aid performance with complex signals. James M. Kates (City Univ. of New York, Graduate Ctr., Rm. 901, 33 W. 42nd St., New York, NY 10036)

New test procedures are being developed for hearing aids based on the response to complex test signals. The tests are motivated by the need to better describe the response of a hearing aid to a speechlike stimulus, since conventional measurements, such as pure-tone sweeps, are often inadequate in showing the behavior that will occur under conditions of actual use. The hearing-aid response typically includes both linear and nonlinear behavior, and both need to be described. Measurement techniques that will be discussed include the use of speech-shaped noise to measure frequency response, the use of bias tones to help determine the frequency response of a nonlinear device such as an automatic gain-control (AGC) circuit, the use of coherence to measure nonlinear distortion, and the use of recorded speech to measure the effects of adaptive signal-processing circuits.

Contributed Papers

9:45

4EA6. A three-dimensional sound intensity probe. Shun Oguro, Masazou Anzai, Hideo Suzuki, and Takahiko Ono (Ono Sokki Co., Ltd., 1-16-1 Hakusan, Midoriku, Yokohama, 226 Japan)

A commonly used sound intensity probe is one-dimensional, that is, only the sound intensity in the direction of the probe axis is measured. This makes the sound intensity measurement very time consuming if two- or three-dimensional intensities are needed. Very rarely a two- or three-dimensional intensity probe is used for practical applications. A conventional three-dimensional intensity probe consists of six (three pair) microphones, making the probe itself very clumsy. Due to its

non-negligible size, the sound field is disturbed and an accurate measurement becomes difficult. A three-dimensional intensity probe was developed that solves this problem. Four 1/4-in. microphones are located at each apex of a regular tetrahedron. Each of them are attached at the tops of four parallel tubes with a 4 mm diameter. When viewed from the front, the four microphones are located at the three apexes and the center of a triangle. The pressure at the center of the tetrahedron is given by the average of the four pressure outputs. The intensities from the center to the four apexes are obtained by use of the average pressure and individual pressure output. These four components are distributed to three (x,y,z) components. Results of numerical calculations that show the accuracy of the algorithm will be given.

10:15

4EA7. Optical techniques applied to underwater surface scattering measurements. H. R. Suiter (Code 2120, Coastal Systems Sta., Panama City, FL 32407-5000)

An acoustic lens was used as a collimator to measure periodic surface backscattering. Techniques to align and focus the acoustic lens are described. The performance is evaluated in the context of classical lens aberration theory and was checked experimentally. Using these methods, an experimenter is able to measure high-frequency backscattering in a constrictive tank environment without interference from target edges or concern that far-field conditions are being compromised. Limitations on the method are discussed. [Work supported by ONR.]

10:30

4EA8. Using acoustical measurements to monitor corn moisture. Paul A. Wheeler, Anush Gopalan, and Scott Budge (Dept. of Elec. Eng., UMC-4120, Utah State Univ., Logan, UT 84322)

The moisture content of grain, such as corn, is usually measured by weighing a sample before and after drying. This paper reports the feasibility of measuring corn moisture on-line by sampling and processing the acoustic signals of corn with different moisture content. Acoustic measurements were made in an anechoic chamber. A condenser microphone connected to an Omnidata Polycorder was used to sample the acoustic signal of corn dropping into a bin. An adaptive filter was used to discriminate corn moisture by comparing frequency information from signals of different corn samples. The results demonstrated the feasibility of using acoustical measurements for on-line monitoring of corn moisture.

10:45

4EA9. Effect of a co-flowing annular outer flow on the flow and acoustics in a porous tube. Sundar Ramamoorthy, Fariborz Khodabakhsh, and Sastry Munukutla (Ctr. for Elec. Power, Tennessee Technol. Univ., Box 5032, Cookeville, TN 38505)

One of the most important aspects of flow in a gas turbine combustor is the cooling airflow introduced through the combustor liner. The co-flowing annular cooling air affects the flow and the acoustic field of the main combustor. A generic study is in progress to study the effect of a co-flowing annular outer flow on the flow and acoustics in a porous tube. This work is an idealization of the actual gas turbine combustor flow. The results generated here will be used to validate the computational codes currently being used by the gas turbine industry to calculate these flow fields. In the present experimental work, a 6-in.-diam tube made out of perforated sheet is located coaxially in an 8-in.-diam outer tube. Airflows in the inner perforated tube, as well as in the annular space between the two tubes. Detailed measurements of the turbulence structure using hot wire anemometry and of the acoustic field using microphone transducers are being made. Effects of parameters such as porosity of the tube, relative areas of annular space and cross section of inner tube, and flow Reynolds number on the turbulence quantities and the acoustic field will be reported.

11:00

4EA10. A general purpose multimedia system for real time transcription of acoustic data to digital storage. Allen E. Leybourne (School of Eng. Technol., Univ. of Southern Mississippi, Southern Station Box 5137, Hattiesburg, MS 39406) and Anthony T. Pogue (Naval Res. Lab., Stennis Space Center, MS 39529)

Activities of the Arctic Acoustic Branch of the Naval Research Laboratory at Stennis Space Center, MS often require transcription of field recorded multichannel acoustic data into digital streams. Insertion of time codes, synchronization words, and check fields into this stream is difficult because of the real time requirement of the transcription process. This paper describes our experience with the integration of components to obtain a multimedia general purpose 2- to 110-channel system whose end product is stored on 2.3 Gigabyte, 8-mm (exabyte) tapes. The system has successfully processed direct analog data, analog data originally recorded on magnetic tape, pulse code modulated (PCM) data tapes, and digital data streaming tapes. Major components of the system are the LORAL System 500, Racal Storehorse, Panasonic Hi-Fi VCR, an in-house developed EVCR (expanded video cassette recorder) system, DATATAPE model DCR8-L5 and a Digital MicroVAX II which continuously receives the LORAL data stream and writes the exabyte tapes. These may then be processed off-line on computers that support the exabyte tape drives. As header information is easily included at the beginning of these tapes, the archival value of this storage format is significantly enhanced.

11:15

4EA11. A maximal-length sequence technique for estimating the coefficients of a pole-zero parametric model of an acoustical system. Paul S. Kovitz and David C. Swanson (Graduate Program in Acoust., Penn State Univ., Appl. Sci. Bldg., University Park, PA 16802)

A technique is described for estimating pole-zero models of real acoustical systems using an excitation signal known as a maximal-length sequence (MLS). The MLS is useful because for any sequence length, the autocorrelation (and corresponding power spectrum) is nearly the same as that for an infinite-length white-noise sequence. The pole-zero model for the system can be used to simulate the response of the actual acoustic system to any input. In addition, a pole-zero filter model can be synthesized from theoretical descriptions of the underlying physics such as Green's functions. Therefore, pole-zero modeling of acoustic systems provides a convenient way to compare theory and experiment. An example is presented consisting of a one-dimensional waveguide excited by a loudspeaker that is driven by a MLS signal.

11:30

4EA12. Real time wave vector filtering in one-dimensional media. Gary P. Gibbs, Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061), and Richard J. Silcox (NASA Langley Res. Ctr., Hampton, VA 23665)

Often it is necessary to measure the wave vector components of propagating waves in vibrating media. Some examples are: measurement of traveling flexural and extensional waves in beams; measurement of acoustic waves in ducts; and determination of boundary conditions in structural or acoustic systems. In previous work, a time domain technique for wave vector filtering of plane waves in ducts was developed which required that the sensor spacing relative to the wavelength of

interest be small. Frequency domain techniques have also been developed to perform wave vector filtering in both dispersive and nondispersive media. These techniques cannot be used in real time because of the signal processing requirements. In this work, a new time domain technique is presented that utilizes two sensor locations in conjunction with a digital filter, for each wave type, to provide a real time estimate of the complex amplitude of each of the traveling wave components for both dispersive and nondispersive media. The technique estimates the wave components over a band-limited spectra whose limits are dictated by the spacing between the sensors (up to a spacing of half a wavelength). A simulation is used to determine the performance of the new technique in both dispersive and nondispersive media over a range of frequencies.

11:45

4EA13. Application of a new signal processing technique in a multibeam sounder. Bishwajit Chakraborty and Y. Satyanarayana Raju (Natl. Inst. of Oceanogr., Dona Paula, Goa 403004, India)

The currently used beamforming method for multibeam sounding application is a delay-sum method, which is also known as a Fourier spectral estimation method applicable to directional estimation. But array size is one major difficulty that exists with this technique. A solution to such a problem is the application of a "high-resolution technique" for the beamforming method. In this paper beamforming techniques like the maximum likelihood method (MLM) and the maximum entropy method (MEM) are presented for multibeam sounding application. The size of the array is chosen to be one-fourth and one-half of the conventional array at an input signal-to-noise ratio of 0 or 10 dB. Both high-resolution techniques are found to be suitable for multibeam sounding use at a 10-dB signal-to-noise ratio (input), even for a small array size of 16 elements. But relatively wide beamwidths for the 45° direction, when the input signal-to-noise ratio is 0 dB, does not support its utility for both techniques. The lowest value of the standard deviation angular error for MEM shows its preference over the MLM technique.

WEDNESDAY MORNING, 13 MAY 1992

WEST END, GRAND BALLROOM CONCOURSE, 9:00 TO 11:30 A.M.

Session 4ED

Education in Acoustics: Undergraduate Projects in Acoustics (Poster Session)

Anthony A. Atchley, Chair

Department of Physics, Code PH/Ay, Naval Postgraduate School, Monterey, California 93943

Contributed Papers

All posters will be on display and all authors will be at their posters from 9:00 to 11:30 a.m.

4ED1. Acoustic related senior projects in Electrical Engineering at Utah State University. Paul A. Wheeler (Dept. of Elec. Eng., UMC-4120, Utah State Univ., Logan, UT 84322)

With the increase in acoustic research in the Electrical Engineering Department at Utah State University comes an increase in the number of acoustic related senior projects. This poster presentation will display several of the senior projects performed. These include such projects as: (1) sound system design software for matching sound system equipment to different performance environments; (2) analog audio synthesizer designs; (3) design of active crossover networks; and (4) audio amplifier/equalizer designs. Including acoustic-related senior projects in the electrical engineering curriculum allows students to apply the theory learned in classes to real-world experiences. It increases a student's interest in the field of acoustics and prepares new engineers for employment in the audio industry.

4ED2. Experimental investigation on the amplification of hydrodynamic noise generation by the insertion of bubbles in a turbulent flow. Charlene E. Hughes and Murray S. Korman (Dept. of Phys., U.S. Naval Academy, Annapolis, MD 21402)

An apparatus is constructed to produce a turbulent shear flow that is generated by a submerged circular jet. The jet is arranged to flow in

an upright position. Comparisons are made of the near-field hydrodynamic flow noise, when the jet flow becomes a composition of two-phase flow that consists of air bubbles in water. The near-field flow noise is measured by a hydrophone located at the point, four nozzle diameters from the exit and four nozzle diameters perpendicular to the jet axis. Preliminary results indicate that the acoustic spectra are greatly amplified in the case where the bubbles are introduced into the flow. Measurements of the acoustic intensity are made as a function of void fraction β (the ratio of air volume to total volume) in an effort to verify the theoretical amplification predictions made by Crighton and Ffowcs-Williams [J. Fluid Mech. **36**, 585-603 (1969)] and more recently by Prosperetti [J. Acoust. Soc. Am. **84**, 1042-1054 (1988)]. Tests are performed in the U.S. Naval Academy Hydrodynamics Tow Tank. [This research is a continuation of the work on ocean noise mechanisms that is supported by the National Center for Physical Acoustics and the ONR.]

4ED3. Exploration of turbulence by nonlinear acoustic scattering. James E. Parker, III and Murray S. Korman (Department of Physics, U.S. Naval Academy, Annapolis, MD 21402)

A system has been developed to explore the nuances of nonlinear scattering in the presence of turbulence. The computer-controlled scattering apparatus makes translational and rotational "crossed beam"

measurements of the nonlinear interaction. This is made possible by the geometry of the transducer setup, resolution of the focused transducers, and mobility of the transducer mounting unit. Two ultrasonic focused transducers (of sinusoidal driving frequencies 2.0 and 2.1 MHz) are mounted mutually perpendicular and with their focal points overlapping. The enhanced spatial resolution results from the minimal interaction volume defined by the intersection of the focused sound beams. The transducer mount is allowed to pivot about the vertical axis while independently translating across a submerged water jet. The $\frac{1}{4}$ -in. circular jet is the source of turbulence in this system. By making several cat scans of the turbulent jet at all angles, one is able to map out several statistical properties of the randomly fluctuating turbulent velocity components. In particular, one can experimentally determine the energy spectrum of turbulence $E(k,r)$ (where k is the wave number of the turbulent eddies and r the radial position in the jet) [M. S. Korman and R. T. Beyer, *J. Acoust. Soc. Am.* **85**, 611-620 (1989)]. [Work supported by Naval Academy Research Council.]

4ED4. Materials characterization via acoustic levitation in air. Bob Boulware and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

A system for acoustically levitating small objects in air at 25 kHz has been built. The system consists of two PZT crystals driving an aluminum horn to produce a vertical standing wave field between the horn end and a curved reflector. Both the solid particles and liquid drops were levitated, and the deformation and motion of these objects were measured in response to an excitation. The detection system con-

sists of a He-Ne laser illuminating the object, which casts its shadow on a photodiode. Light extinction variations cause a fluctuating voltage signal from the photodiode that can be correlated with the motion or deformation of the object in response to impulsive or continuous acoustic excitations. Quantitative information about the mass, viscosity, and surface tension of objects has been derived from this signal for a variety of materials.

4ED5. Auditorium acoustics simulation for the Macintosh. Marc Rieffel and E. Carr Everbach (Dept. of Eng., Swarthmore College, Swarthmore, PA 19081-1397)

Geometrical ray-tracing techniques have been used as educational and analysis tools in a Macintosh program written in Think™ C. This program takes as an input file a standard DXF-format file created from any of several popular CAD programs. From this data, a wire-frame representation of the auditorium is displayed and the user is able to specify the acoustic reflection characteristics of each surface. The program then allows the user to specify the position, frequency characteristics, and directionality of sound sources in the hall, and to run a ray-tracing simulation represented in 3-D color graphics of the behavior of sound rays emanating from the sources. Diffraction effects and air attenuation are accounted for in the propagation algorithm. Sound level contours through arbitrary planes are available after the simulation terminates, as are a variety of acoustic descriptors for user-specified listener locations. Hopefully this normal-mode analysis for steady excitations can be incorporated into newer versions of the program.

WEDNESDAY MORNING, 13 MAY 1992

SALON I, 8:30 TO 10:45 A.M.

Session 4MU

Musical Acoustics: General Topics

E. Paul Palmer, Chair

Department of Physics and Astronomy, Brigham Young University, Provo, Utah 84602

Chair's Introduction—8:30

Contributed Papers

8:35

4MU1. Modal analysis of a Korean bell. Thomas D. Rossing and Alexis Perrier^{a)} (Dept. of Phys., Northern Illinois Univ., DeKalb, IL 60115)

Large temple bells have been cast in Korea for more than 1200 yr. Most Korean bells have a circular area, called a *dang jwa*, which provides a convenient striking point, and a sound pipe, called a *eumtong*, at the top. The asymmetry due to the *dang jwa* and the ornamentation gives rise to mode doublets and results in a slow beating, which is carefully controlled and considered to be an important characteristic of Korean bell sound. The vibrational modes of a small Korean bell have been studied, using holographic interferometry and modal analysis with impact excitation. The vibrational modes are somewhat different from those of carillon bells, tuned church bells, and handbells previously studied [see N. H. Fletcher and T. D. Rossing, *The Physics of Musical Instruments* (Springer-Verlag, New York, 1991), Chap. 21]. No (m, l^*) modes are observed. Mode doublet splittings range from 1% to 9%. The

fundamental (2,0) mode decays much more slowly than the higher modes of vibration, and thus determines the pitch of the bell. [A.P.'s visit was supported by a grant from Ecole Nationale Supérieure des Telecommunications.] ^{a)}Student intern from Ecole Nationale Supérieure des Telecommunications, Paris.

8:50

4MU2. Measurements of nonlinear phenomena in a vibrating wire. Roger J. Hanson, James M. Anderson, and H. Kent Macomber (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50614)

If tensional changes and longitudinal motion of a vibrating string or wire are taken into account, the usual simple theory of vibrating strings is replaced by a complex theory involving coupled nonlinear partial differential equations. The resulting motion is correspondingly complex. Experimental measurements have been made of the motion of a harp-

sichord wire driven electromagnetically in a fixed direction perpendicular to the equilibrium position of the wire. Near resonance the amplitude of the motion perpendicular to the driving direction can become nearly as large as in the driving direction and harmonics of the driving frequency can be pronounced. The outputs of the two orthogonal optical motion detectors were fed to a dual-channel frequency analyzer and an *X-Y* oscilloscope. Measurements were made of frequency, amplitude, and phase of the motions in directions parallel and perpendicular to the driving force. Results will be presented for the nonlinear motion that has been observed for amplitudes of 5 mm down to less than a micron on a wire 0.7 m long. The larger amplitude motions will be exhibited on a video tape.

9:05

4MU3. Multiconvolution in waveguides with arbitrarily spaced discontinuities. Ana Barjau, Salvador Cardona (Dept. d'Enginyeria Mecànica, Univ. Politècnica de Catalunya, Diagonal 647, 08028 Barcelona, Spain), and Douglas H. Keefe (School of Music, DN-10, Univ. of Washington, Seattle, WA 98195)

The multiconvolution algorithm [Martínez *et al.*, *J. Acoust. Soc. Am.* **84**, 1620–1627 (1988)] used to calculate the impulse response or reflection function of a musical instrument air column has proven to be useful, but it has the limitation that the spacing between discontinuities is constrained to be some multiple of $c \Delta t$ (for phase velocity c and time step Δt). Two approaches have been devised to remove this limitation. Each time-domain approach calculates the response of an air column, modeled as an arbitrary one-dimensional acoustic waveguide constructed using cylindrical or conical bore segments with viscothermal

damping and tone-hole discontinuities. The band-limited discrete-time multiconvolution (BDTM) specifies band-limited discrete-time reflection and transmission functions at each discontinuity. The continuous-time interpolated multiconvolution (CTIM) uses continuous-time convolutions between analytical reflection and transmission functions and discrete-time pressure signals. The arbitrary spacing between discontinuities is accounted for in the BDTM method by multirate discrete-time processing of the response functions, and in the CTIM method by interpolation of the discrete-time pressure signals.

9:20

4MU4. Remarks on the reed-air column system as coupled oscillators. Gabriel Weinreich (Randall Lab. of Phys., Univ. of Michigan, Ann Arbor, MI 48109-1120)

The combination of a reed (outward- or inward-striking) and a resonant air column represents a pair of oscillators which mutually influence each other in a non-Hermitian way, allowing one to look for eigenfrequencies of the composite system and for criteria that give their imaginary parts the sign that corresponds to growing oscillations. This approach, although elementary, does not appear to have been used before. The computation is complicated by the fact that the imaginary part is a rather small fraction of the frequency, so that the usual reduction of a quartic to a quadratic secular equation provides at best a first approximation. Experiments are being designed to test these results for the simple case of an air "column" with but a single resonance; it is also expected that these calculations will be helpful in computer simulations of wind instrument systems. [Work supported by NSF.]

9:35–9:45

Break

9:45

4MU5. Octave illusion revisited. Wenyi Huang, Richard E. Pastore, and Michael D. Hall (Psychoacoust. Lab. and the Cognitive and Psycholinguistic Sci. Ctr., Dept. of Psychology, SUNY at Binghamton, Binghamton, NY 13902-6000)

The octave illusion occurs when two tones an octave apart are repetitively presented dichotically in sequence such that when one ear receives the low tone the other ear simultaneously receives the high tone, with each subsequent presentation of the tones switching ear. The illusion is described as perception of a single tone whose location switches from ear to ear and whose pitch switches from one octave to the other in synchrony with the local shift [D. Deutsch, *Nature* **251**, 307–309 (1974)]. The current study investigates various factors that may contribute to the illusion. Initial experiments replicate the illusion, examining temporal parameters critical to streaming and mapping the frequency of the pitches heard in the illusion. Results suggest that subjects hear only small shifts in pitch and lateralized position, with a major factor being fusion of the simultaneously presented stimuli. Later experiments first explored the pitch and location of a single dichotic pair of 400- and 800-Hz tones, then determined the likelihood of fusion for other harmonically related dichotic stimuli. [Work supported by NSF.]

10:00

4MU6. Loudness levels of orchestral instruments. Andrzej Miskiewicz (Auditory Perception Lab., Northeastern Univ., 413

Mugar Hall, Boston, MA 02115) and Andrzej Rakowski (Chopin Academy of Music, Okolnik 2, 00-368 Warszawa, Poland)

The loudness levels of nonpercussive orchestral instruments were calculated from 1/3-octave-band sound pressure levels using Zwicker's method, as well as Stevens' Mark VI and Mark VII procedures. The sounds measured were scale segments played pianissimo and fortissimo in various pitch registers. Calculated for bowed instruments, the bassoon, and the oboe, the loudness of scales performed at a constant playing level is nearly invariant in different pitch registers. In contrast, for the flute, clarinet, and brass instruments, the loudness level markedly increases with increases in pitch. Zwicker's method generally yields higher values of loudness level than both of Stevens' procedures. The differences between Zwicker's and Stevens' phons are more pronounced in the low-pitch registers than in the higher ones. Moreover, due to specific spectral characteristics of musical sounds, for most instruments the dynamic range of loudness predicted by Stevens' procedures is larger than the corresponding range predicted by Zwicker's method. [Research supported in part by NIH Grant No. R01NS07270.]

10:15

4MU7. Determination of musical meter using the method of autocorrelation. Judith C. Brown (Media Lab., MIT, Cambridge, MA 02139 and Wellesley College, Wellesley, MA 02181)

The melodic lines of examples of compositions from the baroque,

classical, romantic, and contemporary periods have been studied in order to determine the musical meter by computer calculation. The method of autocorrelation is appropriate for this calculation since it is a measure of the frequency of occurrence of events following an event at time zero. If a greater frequency of events occurs on the downbeat of a measure as predicted by Palmer and Krumhansl ["Mental Representations for Musical Meter," *J. Exptl. Psychol.: Human Percept. Perform.* **16**, 728-741 (1990)], then a peak in the autocorrelation function should indicate the time for a single measure. The results of these calculations indicate that computer determination of tempo is quite feasible. Autocorrelation results will be presented in graphical form.

10:30

4MU8. A high-resolution fundamental frequency determination based on phase changes of the FFT. Judith C. Brown (Media Lab., MIT, Cambridge, MA 02139 and Wellesley College, Wellesley, MA 02181)

The constant Q transform described recently [J. C. Brown and M. S. Puckette, "A Real Time Constant Q Transform," submitted to *IEEE*

Trans. Signal Process. (1992)] has been adapted so that it is suitable for tracking the fundamental frequency of extremely rapid musical passages. For this purpose a frame size of 25 ms or less is required; thus the calculation described previously has been modified so that it is constant resolution rather than constant Q for the lower frequency bins. This calculation as modified serves as the input for a fundamental frequency tracker similar to that described by Brown [J. C. Brown, "Musical Pitch Tracking using a Pattern Recognition Method," submitted to *J. Acoust. Soc. Am.* (1992)]. This frequency tracker is designed for musical applications and is based on pattern recognition of the log frequency representation of a sound with harmonic frequency components. Once the fundamental frequency has been chosen by the tracker, an approximation for the phase change in the FFT for a time advance of one sample was used to obtain an extremely precise value for this frequency. Graphical examples are given for musical passages by a violin executing vibrato and glissando. Here the instrumental frequency changes are continuous, and an extremely accurate value is necessary for a precise determination of the fundamental frequency.

WEDNESDAY MORNING, 13 MAY 1992

SALON G, 8:30 TO 11:05 A.M.

Session 4NS

Noise: Nonoccupational Noise Exposure

Daniel L. Johnson, Chair

EG&G Special Projects, 2450 Alamo Southeast, Albuquerque, New Mexico 87119

Chair's Introduction—8:30

Invited Papers

8:35

4NS1. Nonoccupational noise exposure: Is it a problem? Daniel L. Johnson (EG&G Special Projects, Biophys. Operation, 2450 Alamo SE, Albuquerque, NM 87119)

If the differences of hearing levels between males and females, as indicated in either the 1960-62 or 1971-75 Public Health Service Reports, are compared with the noise induced permanent threshold shifts (NIPTS) as predicted by ISO1999, these differences fit the NIPTS expected from an exposure of the males to 8 h levels from 89 to 93 dB. Since the number of male workers exposed to such noise is expected to be but a fraction of the total male population, nonoccupational noise exposure must be the most important contribution to this difference. Supporting figures and analysis will be presented to make this point. The role of dosimetry and noise exposure questionnaires in defining nonoccupational noise will also be discussed.

9:00

4NS2. Noise exposure in children and youths. William W. Clark (Central Inst. for the Deaf, 818 South Euclid, St. Louis, MO 63110)

Over the past several years there has been an increasing concern about excessive noise exposure in children and youths. Noise-induced hearing loss, once considered to be an occupational hazard, can be caused by any acoustic environment with sufficient energy to damage the cochlea. This presentation reviews available data on exposure levels and patterns from the most common sources of nonoccupational noise in children: impulse noise from toy cap pistols and firecrackers in children and small-caliber firearms in youths, attendance at rock concerts, listening to amplified music through stereo headphones; and noisy school and leisure activities. For each exposure, consideration is given to the "dose" of noise experienced by the typical consumer, and the distribution of "doses" across users. The risk of each noisy activity causing or contributing to noise-induced hearing loss is determined by considering these factors and the prevalence of the activity in children and youths. Strategies are proposed to educate young consumers about the potential hazards to hearing from these exposures. [Work supported by NIOSH.]

9:25

4NS4. Noise-induced hearing loss in children and adolescents. Patrick E. Brookhouser (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

Occupational noise exposure remains the most commonly identified cause of noise-induced hearing loss (NIHL) in the general population. The issue of NIHL in children has received scant attention. Parents and public policy makers have expressed increasing concern about hearing risks to children and adolescents posed by noise sources such as: rock music, "boom" cars, loud toys, recreational vehicles, and personal cassette players. This report focuses on 114 children and adolescents (ages 19 and under; 90.3% males) who were diagnosed as having probable NIHL on the basis of clinical history and audiometric configuration. Forty-two children had unilateral losses, while 72 had sensorineural losses of varying degree in the contralateral ear. The mean age of referral for evaluation was 12.7 yr (range 1.2 to 19.8 yr, s.d. 4.21 yr), although 26% of the children were age 10 or younger at the time of referral. Inherent weaknesses of currently available clinical studies of NIHL in nonoccupational settings include imprecision in diagnosis and lack of quantification in characterizing the nature and duration of exposure. The occurrence of irreversible but potentially preventable hearing loss among a preschool and school-age sample should prompt improved observational and interventional, prospective longitudinal studies of NIHL in children and adolescents. Advances in molecular medicine as related to hearing loss, incorporated into such studies, could enhance understanding of individual variation in susceptibility to hearing damage from noise associated with age, gender, racial or ethnic status, and genetic make-up. Psychosocial measures should address the issue of improving compliance in the utilization of ear protectors among the younger age group.

9:50-10:05

Break

Contributed Papers

10:05

4NS5. Human auditory response to low-flying aircraft noise. Charles W. Nixon, Denise M. West, and Nancy K. Allen (Bioacoust. and Biocommun. Armstrong Lab./CFBA, Wright-Patterson AFB, OH 45433-6573)

Noise from low-flying aircraft may constitute a potential risk of noise-induced hearing loss. In this study, hearing threshold levels were measured following human exposures to audio recordings at ground level of noise from low-flying aircraft. The noise was characterized by sudden onset, durations of about 10 s, maximum A-weighted sound pressure levels in the frequency region of 1200-2400 Hz, and A-weighted peak levels up to 130 dB. The primary independent variables were A-weighted peak levels and successive repetitions of the noise. In phase I, subjects experienced single noise bursts at a maximum level of 130 dB with no observable changes in hearing, as reported at the ASA meeting in Houston. In phase II, a single noise burst was repeated at the same sound level up to eight successive repetitions or until a TTS was observed. Hearing threshold levels were measured at 4 and 6 kHz immediately following each individual noise burst. The results of phase II and the overall study will be discussed in terms of noise-induced hearing loss relative to damage risk criteria and ISO 1999.

10:20

4NS6. Geographic representation of noticeability of aircraft noise in Grand Canyon National Park. Sanford Fidell, Matthew Sneddon,

John Smyth, and Karl Pearsons (BBN Systems and Technol. Corp., P.O. Box 633, Canoga Park, CA 91304)

A novel geographic representation of the noticeability of aircraft noise that accounts for topographic relief has been constructed in a distributed computing environment. A regular grid of points spaced at 300-m intervals was superimposed over Grand Canyon National Park. Starting from digital elevation model data, GRASS geographic information system software running on a Sun workstation determined line-of-sight transects from each surface grid point to points on a complex system of aircraft flight paths. An acoustic propagation model based on algorithms of the U.S. Army Tank-Automotive Command's ADRPM-7 software was then exercised on networked VAX and Butterfly computers to predict 1/3-octave-band aircraft noise levels produced by operation of a fleet of aircraft at each grid point. Bandwidth-adjusted signal-to-noise ratios were then calculated and returned to the Sun for construction of a new GRASS map layer of noticeability levels for the park. The new map can be color coded, draped over terrain contours, and rendered in perspective views for visualization purposes.

10:35

4NS7. New facility for very-low-frequency acoustic testing. Andrew Kugler and Matthew Sneddon (BBN Systems and Technol. Corp., P.O. Box 633, Canoga Park, CA 91304)

A very-low-frequency test facility has just been completed for experimentation on the effects of impulsive and other low-frequency noise

exposure on structures, animals, and people. The facility is a cube of steel-reinforced concrete with outside dimensions of 4.6 m (wide) by 6.7 m (long) by 3.2 m (high). The interior volume can be partitioned into test volumes ranging from 3.6 to 32.4 m³. Sound pressure levels as great as 155 dB can be generated between 0.5 to 100 Hz by 72 servomotor-driven transducers. Computer-generated test signals of arbitrary shape and duration can be produced, and a fully automated data acquisition and analysis system is available to aid efficient experimentation.

11:00

4NS8. Assessment of the volume of environment noise SNL CNL and their utilization. Lin Cong (Inst. of Environmental Science, Fujian Province, People's Republic of China)

In this paper the author has analyzed the relation between human subjective sense and various environmental noise level indexes and conducted a comprehensive study of the correlation between different sound levels and subjective annoying rates using the principle of applied mathematics and experimental psychology. Comprehensive indexes in sound environmental assessment, SNL and CNL, are introduced and the method of limited isolated sound level indexes has been changed into the calculation of psychological effect of sound over humans as assessment volume in environmental noise assessment so that the assessment index can reflect directly psychologically acceptable limits of humans to surrounding sound. This shows the percentage of people affected by noise under the surrounding sound. Meanwhile, the author also gives a detailed description of the application of the assessment index to the analysis and prediction of sound environmental quality, of construction project planning, and of the advantages of the assessment index.

WEDNESDAY MORNING, 13 MAY 1992

SALON B, 8:25 A.M. TO 12:00 NOON

Session 4PA

Physical Acoustics: Sound Attenuating Materials

Jacek Jarzynski, Cochair

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

Robert D. Corsaro, Cochair

Code 5135, Naval Research Laboratory, Washington, DC 20375-5000

Chair's Introduction—8:25

Invited Papers

8:30

4PA1. Loss factor height-width tradeoffs for viscoelastic materials. Bruce Hartmann, Gilbert F. Lee, and John D. Lee (Polymer Phys. Group, NSWC, Silver Spring, MD 20903-5000)

It is observed empirically that the height of the dynamic mechanical shear loss factor peak in a viscoelastic material is inversely related to the width of the peak (as a function of frequency). There is a question of whether this observation represents a fundamental limitation of nature or whether some new material might be found with a peak that is both high and broad. Starting with a single relaxation time model and progressing to the Havriliak-Negami model, the tradeoffs between height and width are pointed out. These analytical models confirm that the empirical observations represent fundamental limitations that high peaks are narrow and low peaks are broad. The height of the peak is maximized as the material approaches a single relaxation time. The width is maximized by minimizing the ratio of the relaxed modulus to the unrelaxed modulus and by minimizing the asymmetry of the relaxation. For realistic values of the parameters, a loss factor peak height of 2 can have a width of no more than about 3 decades. Conversely, a relaxation with a width of 8 decades can have a loss factor peak height of no more than 0.5.

9:00

4PA2. A group contribution analysis of the integral of the linear loss modulus-temperature curves. L. H. Sperling (Mater. Res. Ctr., Ctr. for Polymer Sci. and Eng., Dept. of Chem. Eng. and Mater. Sci. and Eng., Lehigh Univ., Bethlehem, PA 18015)

The integral of the linear loss modulus-temperature curves, termed the loss area, or L.A., is controlled in part by the chemical structure of the mers making up the polymers being examined. Another contribution arises through the morphology of multipolymer materials, particularly phase domain continuity. Research to date has centered on acrylic, styrenic, and vinyl polymers and interpenetrating polymer networks made from these polymers. The relationships among chemical structure, morphology, and damping capability will

be discussed. The requirements for instrumental background corrections to the data will be examined, and several proposed corrections will be compared. Possible methods of maximizing (or minimizing!) damping will be explored. [Work supported by ONR.]

9:30

4PA3. Viscoelasticity of crosslinked gels. Douglas Adolf and James E. Martin (Sandia Natl. Labs., Physical Properties of Polymers Div., P.O. Box 5800, Albuquerque, NM 87185)

The damping characteristics of a crosslinked polymer are determined by its viscoelastic properties. Yet the loss mechanisms of networks have not been fully understood. It is shown that insight into network behavior can be gained from studies of how viscoelasticity *evolves* during the curing process. In addition, the peculiar viscoelasticity of the incipient gel (the polymer exactly at the gel point) is shown to offer unique damping characteristics. Experimental studies on an epoxy system show that the evolution of structure prior to the gel point follows the predictions of percolation and that the divergence of the bulk viscosity and the power law behavior of the dynamic shear moduli, $G' \sim G'' \sim \omega^\Delta$, at the gel point follow the predictions of a dynamic scaling theory. A new superposition principle (time-cure superposition) holds for the linear viscoelastic responses at different extents of reaction in the critical regime both before and after the gel point. From the master curve beyond the gel point, the divergence of the inverse equilibrium modulus and the terminal relaxation time are shown to obey the theoretical predictions as well. In addition, an intriguing relationship exists between the relaxation time spectra of a network at the gel point and in the fully cured state. Measurements of the dynamic shear moduli of several epoxy and siloxane networks, with equilibrium moduli varying by four orders of magnitude, show $G'' \sim \omega^\Omega$ at all extents of reaction past the gel point, where the network viscoelastic exponent Ω remains at the incipient gel value Δ . This surprising correspondence suggests that chemical networks possess some remnant of the fractal structure of the incipient gel. [Work supported by the U.S. DOE under Contract No. DE-AC-04-76DP00789.]

10:00

4PA4. Acoustic properties of materials with reduced Poisson ratio. Subhash C. Narang and Asutosh Nigam (SRI Intl., 333 Ravenswood, Menlo Park, CA 94025)

Reduced Poisson ratio materials present an interesting opportunity for altering acoustic properties. Relationship of material morphology with Poisson ratio will be explored. The effect of Poisson ratio on mechanical properties will be reviewed. Relationship with moduli, flexural rigidity, and fracture toughness will be examined. Reduced Poisson materials have low cutoff frequencies and low wave dispersion constants compared to analogous conventional materials. Development of a variety of materials for specific applications will be discussed. The generic approach to the fabrication of reduced Poisson ratio materials, such as foams, honeycombs, laminates, and composites, provides application engineers added flexibility in choosing appropriate materials.

10:30

4PA5. Modeling and analysis of echo reduction mechanisms in a composite material containing inclusions. Roger H. Hackman (Lockheed Palo Alto Res. Labs., Palo Alto, CA 94304-1191)

The sound attenuating mechanisms of a class of composite materials containing dense inclusions are analyzed. The basis for this analysis is the calculation of the elastic wave scattering from a single spherical inclusion embedded in a microinhomogeneous medium. The role of the rigid body translational mode of the inclusion and its relationship to PS conversion in the coating are described. Multiple scattering effects and the potential role of collective modes in a many inclusion system are also discussed. A parametric T -matrix analysis of the scattering addresses the potential for tailoring the frequency response of the coating through variation of the material properties of the inclusion and of the host medium within allowable constraints. The effects of varying the inclusion shape are likewise addressed by a spheroidal coordinate based T -matrix analysis of the scattering from prolate spheroidal inclusions of varying aspect ratio. Recent improvements in the treatment of multiple scattering aspects of this problem will also be reviewed.

11:00

4PA6. Attenuation of sound in porous materials. Thomas J. Plona (Schlumberger-Doll Res., Old Quarry Rd., Ridgefield, CT 06877)

This paper gives a brief review of the attenuation properties of porous materials when the pore space is filled with either liquids or gases. Liquid-saturated materials (i.e., rocks, sediments, synthetics) will be shown to exhibit three primary mechanisms of sound attenuation: local flow, global flow (i.e., Biot theory), and scattering. How these mechanisms depend on liquid and solid properties (frame stiffness), porosity, pore shape, fluid permeability, pressure, and temperature (etc.) will be emphasized. Attenuation in air-saturated materials (i.e., soils, snow, synthetics) will be discussed in a similar manner with an emphasis on

the nearly rigid nature of the solid. The unique two compressional wave behavior of porous media (fast and slow waves or type I and type II waves) will be reviewed as well as their history from Rayleigh to Biot.

Contributed Papers

11:30

4PA7. A robust transfer matrix formulation for ultrasonic wave propagation in multilayered absorbing media. Daniel Lévesque and Luc Piché (Industrial Mater. Inst., Natl. Res. Council of Canada, 75 De Mortagne, Boucherville, Québec J4B 6Y4, Canada)

An improved version of the transfer matrix approach is presented where general expressions are obtained for reflection and transmission coefficients in either fluid or solid half-space and problems associated with numerical stability are solved efficiently. The formulation is applicable to longitudinal and shear input wave alike, at arbitrary incidence angles and for any sequence of solid or fluid layers. Also, allowance is made for viscoelastic behavior by means of relaxation functions and the response to arbitrary incident pulse shapes and beam profiles is obtained through a two-dimensional numerical Laplace inversion. For application to bond quality inspection, the interface between any two layers is handled by introducing a very thin interphase layer, rigidly bonded to its neighbors, with adjustable viscoelastic properties. In the case of a metal-polymer-metal structure, simulations show resonance splitting, which measures coupling strength between layers and is found to be very sensitive to small changes in the interfacial conditions. These theoretical results are confirmed by experiments and good estimates are given for the specific stiffness of the interface region.

11:45

4PA8. Compressional waves in an anisotropic porous material: CELCOR. Chaur-Jian Hsu and Thomas J. Plona (Schlumberger-Doll Res., Old Quarry Rd., Ridgefield, CT 06877)

Ultrasonic experiments were made to study the compressional waves that propagate in a fluid-saturated, porous, anisotropic material, CELCOR, made by Corning. This material can be described as a regular packing of hollow cylinders, each with a square cross section. The material is anisotropic, both in the plane containing the cylinder axis and in the plane perpendicular to the cylinder axis. This cell structure has square openings of typically 0.05 in. and ceramic wall thicknesses of 0.01 in. The porosity of the system is about 70%–80%. Along the cylinder axis, two compressional waves are observed, one traveling near the fluid speed and one traveling near the ceramic speed. In the plane perpendicular to the cylinder axis, the compressional wave has an angular velocity variation consistent with the square-lattice nature of the cross section. Ultrasonic results of velocity and attenuation (centered at 500 kHz) will be shown for the cases when the material is saturated either with water or air. The two compressional wave behavior parallel to the cylinder axis will be interpreted in terms of the Biot theory.

WEDNESDAY MORNING, 13 MAY 1992

SALON F, 8:10 A.M. TO 12:00 NOON

Session 4PP

Psychological and Physiological Acoustics: Otoacoustics Emissions, Loudness, and Noise

Jozef J. Zwislocki, Chair

Institute for Sensory Research, Syracuse University, Syracuse, New York 13244-5290

Chair's Introduction—8:10

Invited Paper

8:15

4PP1. An update on the field of otoacoustic emissions. Brenda L. Lonsbury-Martin (Dept. of Otolaryngology, Univ. of Miami Ear Institute, P.O. Box 016960, Miami, FL 33101)

Otoacoustic emissions or sounds generated by the cochlea have been known for almost 15 yr. In that time, the beneficial features of using emissions as response measures of hearing function have been well-realized. Today, the field of otoacoustic emissions has expanded to include three major areas of study that can be described as basic, clinical, and technical applications of the technique. The primary purpose of this presentation is to provide an update on the contributions that otoacoustic emissions have made to our understanding of the basis of cochlear transduction, the screening of hearing and the differential diagnoses of hearing problems, and the straightforward monitoring of peripheral processes including the functional integrities of the acoustic reflex, "cochlear amplifier," and cochlear-efferent system. The relevance of the work of Eberhard Zwicker and his colleagues on otoacoustic emissions will be used as a framework within which to discuss the current status of otoacoustic emissions in the hearing sciences. [Work supported by NIH.]

8:45

4PP2. The possible relationship of longitudinal basilar membrane stiffness to the distribution of spontaneous otoacoustic emission frequencies. C. Talmadge and A. Tubis (Dept. of Phys., Purdue Univ., West Lafayette, IN 47907)

Eberhard Zwicker and his co-workers have focused attention on the characteristic frequency spacing (≈ 0.4 Bark) of families of very narrow-band spontaneous otoacoustic emissions (SOAE's). To date, the only type of cochlear model that has been used to describe such SOAE families is the *ad hoc* one in which negative damping with stabilizing nonlinearity is localized so as to correspond tonotopically to the SOAE frequencies. Another type of model is proposed for these SOAE families based on their analogy with the narrow-band spectra of wind and bowed-string musical instruments resulting from self-sustained oscillations that reflect the normal modes of the underlying vibrating air column or string. It is assumed that the basilar membrane has longitudinal stiffness (at least at low levels of displacement), and a study was done to determine to what extent the place dependence of the longitudinal wave velocity can be chosen so as to give membrane modes with the characteristic SOAE frequency spacings. The effect of adding such longitudinal coupling to the usual nonlinear-active damping is explored formally and via numerical simulations. [Work supported by NIH-NIDCD Grant No. DC-00307.]

9:00

4PP3. Can the loudness function determine the shape of the Weber function? William S. Hellman (Dept. of Phys., Boston Univ., Boston, MA 02215) and Rhona P. Hellman (Dept. of Psychol., Northeastern Univ., Boston, MA 02115)

A recent model relating measures of loudness and intensity discrimination [W. S. Hellman and R. P. Hellman, *J. Acoust. Soc. Am.* **87**, 1255-1265 (1990)] is shown to reveal that the near threshold slope of $+1$ obtained for the loudness function requires the near threshold slope of the Weber function for intensity discrimination to be -1 . This result is consistent with psychophysical intensity discrimination measurements for pure tones and broadband noise, as well as with theoretical considerations which predict that the Weber function should have a slope of -1 as the detection threshold is approached [N. F. Viemeister, *Auditory Function*, 213-241 (1988)]. The model also generates the overall shape of the Weber function over a suprathreshold range of sound pressure levels up to 80 dB. Empirical examples are provided. [Partially supported by the Rehab. R. & D. Service of the VA.]

10:00

4PP6. Loudness levels of three complex stimuli and model predictions. Patricia S. Jeng (Ctr. for Res. in Speech and Hear. Sci., City University of New York, New York, NY 10036), Joseph L. Hall (Acoust. Res. Dept., AT&T Bell Labs., Murray Hill, NJ), and Harry Levitt (Ctr. for Res. in Speech and Hear. Sci., City University of New York, New York, NY)

9:15

4PP4. Selective attention to loudness, ignoring pitch. Robert D. Melara (Dept. of Psychol. Sci., Purdue Univ., West Lafayette, IN 47907), Lawrence E. Marks (John B. Pierce Foundation), and Noriko Yamagishi (Purdue University)

Perceptual and decisional separability between the dimensions of loudness and pitch were evaluated in discrimination procedures in which subjects focused on loudness and ignored variation in pitch. In one task, subjects decided on each trial whether a test signal was the same as or different from a referent in loudness. Results showed that subjects were unable to ignore changes in pitch, even though it was in their best interest to do so. Both sensitivity and bias were affected adversely by irrelevant pitch. In a second task, analogous to Garner's speeded classification task, subjects decided on each trial whether a single signal was loud or soft, ignoring trial-to-trial changes in pitch. Under these conditions the dimensions were separable, both perceptually and decisionally; subjects' classification performance was optimal, in the sense that they could ignore pitch completely when it varied orthogonally with loudness, but could integrate pitch when it correlated with loudness. Taken together, the results indicate that successful auditory selective attention is task dependent. The concept of separability is apparently not general to a pair of dimensions, but specific to the conditions under which selective attention is tested.

9:30

4PP5. Binaural loudness summation: Better and less than "perfect." B. E. Mulligan (Dept. of Psychol., Univ. of Georgia, Athens, GA 30602)

Monaural and binaural loudness functions of 500- and 1000-Hz pulsed tones were obtained by loudness estimation under the same conditions where binaural loudness summation previously was found to vary as a function of induced adaptation [B. E. Mulligan *et al.*, *Proc. Int. Soc. Psychophys.* 1-5 (1991)]. Comparisons of loudness function parameters indicate that slopes of all functions tend to be greater under conditions of induced adaptation, but differences between monaural and binaural slopes do not appear to be statistically reliable. Under adapted conditions, binaural/monaural loudness ratios and binaural summation levels exceed "perfect" doubling (ratio of 2 and level of about 10 dB). Under nonadapted conditions, however, loudness ratios and summation levels are less than "perfect" (ratios of approximately 1.4 and levels of about 5-6 dB).

9:45-10:00

Break

Loudness levels were measured for three test stimuli (speech, narrow-band noise, and square wave) at three levels in three test conditions (test stimulus in quiet, test stimulus in presence of masker, and total loudness of test stimulus plus masker). Loudness levels were measured in the traditional way by matching loudness of the test stimulus to that of a 1-kHz tone. In addition, loudness levels were measured using a narrow-band noise as the reference stimulus. The bandwidth of the

narrow-band noise used as test stimulus (NBN-150) was 150 Hz, and the bandwidth of the narrow-band reference stimulus (NBN-120) was 120 Hz. Both stimuli were arithmetically centered at 1 kHz. The loudness level functions for speech and for NBN-150 have slopes of approximately unity. However, the loudness level function for the square wave has a slope of greater than unity below about 50 dB SL and the slope decreases above about 50 dB SL. Loudness predictions of two models are computed and compared to the experimental measurement. [Work of P. S. J. done as a consultant at AT&T Bell Laboratories, Acoustics Research Department.]

10:15

4PP7. Modifications to nonlinear processing for recruitment of loudness. Janet C. Rutledge and Apichat Tungthangthum (Elec. Eng./Comput. Sci. Dept., Northwestern Univ., Evanston, IL 60208)

The time-varying frequency-dependent processing (TVFD) [J. C. Rutledge and M. A. Clements, Proc. IEEE ICASSP, 3641-3644 (1991)] was designed to adjust to changing speech input on both time and frequency basis. Due to inherent problems with the processing technique, some spectral smearing occurs [J. Acoust. Soc. Am. 89, 1974 (A) (1991)]. The processing has been modified to maintain the time-varying frequency-dependent nature, but reduces the smearing effect. A sinusoidal speech model is used with the compensation processing. Appropriate gains for the sinusoid amplitudes are calculated every 5 ms. In the modified processing scheme, the gain determined for the largest peak in each frequency band is applied to all components in that band. This maintains the relative amplitude ratio between neighboring peaks in the frequency spectrum. The compensation algorithm is presented here along with preliminary performance results based on the objective measures described in another paper at this meeting.

10:30

4PP8. Hearing based analysis of engine noise. Gerold Fritz (Neutrik USA, 195 Lehigh Ave., Lakewood, NJ 08701-4527)

Sounds, produced by different car engines were examined with various analysis techniques. Additionally, a group of randomly chosen tests subjects estimated the loudness impression of different sounds. These studies showed, that time function, FFT spectrum, and third-octave spectrum provide no sufficient signal characterization. Different types of car engines, like diesel or otto cycle engines sound different, even though they may have the same spectra. As an alternative a psychoacoustic method is presented, which calculates the impulse content of sound, based on the spectrum of specific loudness. These loudness spectra are calculated according to the "Zwicker standard" directly from the signal in the time domain. The calculated results correlate with the statements of the test subjects very well.

10:45

4PP9. The use of frequency domain kurtosis in the assessment of industrial noise environments for hearing conservation purposes. William A. Ahroon, Sheau-Fang Lei, Robert I. Davis, and Roger P. Hamernik (Auditory Res. Lab., SUNY, Plattsburgh, NY 12901)

Industrial noise environments are frequently characterized by combinations of noise impacts superimposed on a steady-state continuous noise background. Digital techniques were used to synthesize complex noise environments containing random impulses with random peak SPL's up to 125 dB. Four groups of chinchillas were exposed to one of three different complex noise environments whose sound exposure levels and spectra were similar to a fourth Gaussian noise exposure. Conventional methods were used to measure energy spectra and exposure levels. Measures of permanent threshold shift showed significant differences between the complex noise exposures and the Gaussian exposure. An application of frequency domain kurtosis [R. F. Dwyer, IEEE J.

Oceanic Eng. OE-9, 85-92 (1984)] emphasized the region of the noise spectrum where audiometric differences were measured. [Work supported by NIOSH.]

11:00

4PP10. The hazard of exposure to impulse noise as a function of frequency. James H. Patterson, Jr. (USAARL, P.O. Box 577, Fort Rucker, AL 36362-5292), R. P. Hamernik, and W. A. Ahroon (Auditory Res. Lab., SUNY, Plattsburgh, NY 12902)

Existing criteria for safe exposure to impulse noise do not consider the frequency spectrum of an impulse as a variable in the evaluation of the hazards to the auditory system. This report presents the results of a study that was designed to determine the relative potential that impulsive energy concentrated at different frequencies has in causing auditory system trauma. One hundred and thirty (130) chinchilla, divided into 22 groups with 5 to 7 animals per group, were used in these experiments. Pre- and post-exposure audiograms were measured on each animal using avoidance conditioning procedures. Quantitative histology (cochleograms) was used to determine the extent and pattern of the sensory cell damage. The noise exposure stimuli consisted of seven different computer-generated narrow-band impulses (approximately 400-Hz bandwidth) having center frequencies located at 0.260, 0.775, 1.025, 1.350, 2.075, 2.450, and 3.550 kHz. Each narrow-band exposure stimulus was presented at two to four different intensities. An analysis of the audiometric and histological data allowed an iso-hazard weighting function to be derived. The weighting function clearly demonstrates that equivalent amounts of impulsive energy concentrated at different frequencies are not equally hazardous to the auditory system. Comparison of the derived weighting function with the A-weighting curve indicates low frequency impulses are less hazardous than predicted by A-weighted sound exposure level.

11:15

4PP11. The effects of high-level blast wave exposure on hearing in the chinchilla. Roger P. Hamernik, William A. Ahroon, and Robert I. Davis (Auditory Res. Lab., SUNY, Plattsburgh, NY 12901)

A summary of audiometric and histological results from 423 chinchillas that were exposed to high-level (150 to 160 dB peak SPL) blast waves produced by four different sources will be presented. The four sources, in an anechoic environment, produced Friedlander waveforms whose A-weighted energy spectra peaked at 0.25, 1.0, 2.0, or 4.0 kHz. The experimental data were organized to illustrate the effects of sound exposure level (SEL) and spectral energy on auditory system trauma. SEL was increased through increases in peak SPL and number of impulse presentations. The results show that the rate at which pathology (audiometric or histological) grows with increasing SEL is strongly influenced by both the different spectra of the four sources and the spectral details of an individual source. The results can be interpreted to indicate that, for a given octave band, equivalent amounts of energy do not produce equivalent auditory effects but are influenced by the dominant energy-carrying band of the blast wave. [Work supported by USAARL.]

11:30

4PP12. Recovery of auditory threshold following exposures to a pure tone and white noise. I. M. Young, L. D. Lowry, and H. Menduke (Dept. of Otolaryngology, Jefferson Medical College of Thomas Jefferson Univ., Philadelphia, PA 19107)

Recovery from temporary threshold shift (TTS) was measured for a subject with normal hearing following exposures to a continuous pure tone 1000 Hz and white noise. The intensity of the exposure tone was 123, 117, and 110 dB SPL and the exposure duration was 21, 10, and 10 min, respectively. The intensity and duration of exposure to white noise

were 123 dB SPL and 30 min. Automatic audiometer was used to obtain TTS and recovery at a frequency 1500 Hz measured from 1 min to 51 days, 6 days, and 23 h after exposures to 1000 Hz. For white noise, TTS and recovery were measured from 1 to 25 min. Results indicated that, (1) there was steady progression of recovery to pre-exposure threshold, and (2) recovery was neither a simple linear nor monotonic process. These results were compared with our previous study of TTS and recovery obtained at various frequencies following exposures to pure tones 250 and 1000 Hz.

11:45

4PP13. Temporary threshold shift in organists' hearing. Laura E. Brock¹ and Igor V. Nábelek (Dept. of Audiology and Speech Pathology, Univ. of Tennessee, Knoxville, TN 37996-0740)

The purpose of the study was to determine if organists experience temporary threshold shifts of hearing during a typical practice session. Ten organists (6 male, 4 female) aged 23–54 yr were evaluated with hearing history questionnaire, otoscopic examination, otologic questionnaire, and pure tone audiometry. Bone conduction testing verified no conductive hearing loss in any subject. After audiometric examination each organist played the organ for 2 h under identical practice conditions and then was retested with pure tones to assess temporary threshold shift (TTS) in hearing. There was no significant TTS when data were statistically analyzed, however TTSs as large as 15 dB were recorded in individual cases. Spectral content of the organ music played in the sessions was found to have a dominant frequency at around 500–750 Hz.¹ Present address: Dept. of Audiology, Pinehurst Surgical Clinic, P.A., P.O. Box 2000, Pinehurst, NC 28374.

WEDNESDAY MORNING, 13 MAY 1992

SALON H, 8:00 A.M. TO 12:00 NOON

Session 4SA

Structural Acoustics and Vibration: Wide Band/Short Pulse Signal Interaction with Submerged Structures I

Leopold B. Felsen, Chair

Department of Ocean Engineering, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Invited Papers

8:00

4SA1. Observables in short pulse scattering and their relation to structural features. Leopold B. Felsen (Dept. of Elec. Eng., Polytech. Univ., Farmingdale, NY 11735 and Dept. of Ocean Eng. (visiting), MIT, Rm. 5-211, Cambridge, MA 02139)

Short pulse (SP) scattering from submerged structures is related to structural features that leave their imprint via observables in the scattered signature. Because of the SP high resolution, some of the time-domain (TD) observables are isolated spikes and dips while others exhibit quasiperiodic or periodic oscillations. To relate the data to the structure that generates it, as required for modifying the response or for classification and identification, it is necessary to parametrize the scattering process in terms of relevant wave phenomena in the configurational (space-time) as well as the spectral (wave-number–frequency) domains. This can be done by systematic construction of a catalog of observables due to configurations with different structural features. This review summarizes SP results from various relatively simple test problems, with emphasis on configurational, spectral, and hybrid wave objects that are useful for synthesis as well as data processing. Such wave objects include TD edge-diffracted fields, TD leaky and Floquet modes (both traveling and resonant), etc., which can be combined into a hybrid TD ray-acoustic observable-based algorithm. [Work supported by ONR.]

8:25

4SA2. Acoustic field visualizations using short pulse/wide band excitation. William K. Blake and David Feit (Ship Acoust. Dept., David Taylor Res. Ctr., Code 1905, Bethesda, MD 20084)

Acoustic fields radiated and/or scattered by elastic structures are visualized in a number of different ways. This presentation illustrates two possible formats using wide band/short pulse excitation of simple idealized structural configurations. First, the field radiated by a fluid-loaded elastic plate excited by a localized transient force is investigated. The field is calculated using the classical Bernoulli–Euler equations

for the plate response and then compared to results obtained using the Timoshenko–Mindlin plate equations. The transient response is visualized by color coding the pressure levels and displaying them in either rectangular or polar formats, using angle of observation and time as the coordinates. With the fluid sound speed as a reference, precursors were observed in both cases, the nature of which depends on the observation angle and the plate theory used. Second, images are discussed that were obtained from processing returns of wide band acoustic pulses scattered by a body as it rotates relative to the source. The rotation angle determines the lateral, or cross-range resolution, while the signal bandwidth determines the down-range resolution by revealing time domain separation. Alternative images show rotation-angle/down-range loci or cross-range/down-range loci. The displays divulge the existence and location of different classes of structural acoustic phenomena. For a detailed study of the scatterers structure, a joint time-frequency representation at fixed body angles can reveal some inherent features that do not stand out in a single domain representation.

8:55

4SA3. Time domain processing in mid-frequency structural acoustics. A. B. Baggeroer, H. Schmidt, J. R. Fricke, and J. Bondaryk (MIT, Cambridge, MA 02139)

The processing and analysis of high-frequency scattering and radiation experiments have naturally been performed in the time domain, with the sources being short duration transients. This is due to the fact that the scattering and radiation from structural features are easily separated and identified in the time domain. In low-frequency experiments the acoustic wavelengths are large compared to the structure and the structural details in particular. Therefore the analysis and processing have here been naturally performed using frequency domain modal approaches. The mid-frequency regime ($ka \approx 5-20$) provides a transition zone where none of the traditional processing approaches are particularly suitable. Thus, the duration of transients in this regime is longer than the travel time across the structure, making direct time separation of individual arrivals impossible. However, this is a problem encountered in other areas involving wave propagation, including underwater acoustics and seismology. Here a number of high-resolution processing schemes have been developed over the last couple of decades, capable of isolating individual scattering and radiation events, in turn allowing for identification of scattering “hot-spots,” etc. Here, state-of-the-art frequency and time domain signal processing is reviewed, and the merits of the various approaches are discussed. Examples will be given illustrating the use of such high-resolution time domain processing schemes in mid-frequency structural acoustics.

9:25

4SA4. Frequency-time and wave-number-space processing tools for experimental studies of wave propagation in shells. Earl G. Williams (Naval Res. Lab., Code 5137, Washington, DC 20375)

Experimental measurements on point-driven, fluid-loaded cylindrical shells in the frequency region $0 < ka < 11$ are processed using near-field acoustical holography and wave vector filtering in order to expose the details of wave motion on the structure. Signal processing tools along with the creation of various filters in the $k-\omega$ space domain are used to study impulsive wave packets as a function of space and time excited by a point drive inside the shell. This technique provides a very powerful tool for the study of experimental data. Special emphasis will be given to the frequency range that has not presented before, $4 < ka < 11$. Significant is the evidence of shear and longitudinal waves for many circumferential orders. Signal processing methods are presented that include generalized time frequency representations (GTFR) used in the $k-z$ domain, which provide the ability to display wave conversion at discontinuities, such as the ends of the shell. Of particular importance in this frequency range is the radiation from the near field of the point driver, as the WVF processing dramatically shows.

9:50–10:00

Break

10:00

4SA5. Estimation of plate-mode transfer functions near discontinuities via radon transform decomposition. J. Robert Fricke (Dept. of Ocean Eng., Rm. 5-218, 77 Massachusetts Ave., Cambridge, MA 02139)

This paper considers the estimation of complex coupling coefficients, or transfer functions, that relate an incident plate wave encountering an impedance discontinuity to the resulting scattered waves. The discontinuity may be modeled as a linear-time-invariant (LTI) filter while the incident and scattered wave represent the filter input and output, respectively. Using the radon transform, the input and output plate modes are isolated in p -tau space, which facilitates the computation of the transfer function via Fourier transforms and complex division. An example is discussed showing the flexural-to-flexural transfer function that occurs at the end of a truncated steel plate. [Research supported by ONR.]

10:15

4SA6. Isolated feature extraction from composite short pulse scattering data for a thin submerged elastic plate. Tarun Kapoor and Leopold B. Felsen (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Robust interpretation of observables in acoustic scattering data from submerged elastic structures requires decomposition of composite effects into interacting isolated effects, each of which is matched as well as possible to the relevant wave dynamics. Having selected what one believes to be an observable-based parametrization (OBP), one may then proceed to process the data so as to link the observable features with the wave objects that synthesize them. This strategy is applied here to two-dimensional scattering of a line-force-excited short pressure pulse by a submerged thin elastic plate of finite width. The database is generated numerically by a time domain (TD) finite difference code, with observers located on a line parallel, and exterior to, either side of the plate. The selected OBP involves hybrid combinations of ray acoustic geometrically reflected and edged diffracted fields, with coupling to the supersonic (leaky) compressional mode in the plate, and with collective responses due the multiple interactions between the plate edges. The TD processing is built around auxiliary numerical problems that involve a semi-infinite plate and an infinite plate which, respectively, isolate single-edge diffraction and geometrical reflection. By subtracting these numerically from the composite data, one may isolate the TD edge diffraction coefficient, the leaky mode detachment coefficient, and other OBP quantities. The processing is also performed in the radon transformed time-slowness domain. [Work supported by ONR.]

10:30

4SA7. Effects of departures from periodicity on the frequency domain and short pulse acoustic scattering from finite soft and rigid flat strip arrays. L. Carin, T.-T. Hsu, and Leopold B. Felsen (Dept. of Elec. Eng./Weber Res. Inst., Polytech. Univ., Farmingdale, NY 11735)

Many structures of interest contain regularly spaced elements that leave their imprint on the acoustic scattered field. For periodic arrangements, even a few elements may imprint on the scattered field the signature of the periodic structure Floquet modes that characterize an infinite array [L. Carin and L. B. Felsen, *J. Acoust. Soc. Am.* **90**, 2355(A) (1991)]. Explored here, for various collections of flat, soft, and rigid strip scatterers, is how departures from periodicity due to array truncation, as well as displacement, change in size, etc., of indi-

vidual and groups of elements affects the time-harmonic and short pulse plane wave scattered response. Reference solutions—obtained numerically by direct integration of a (spectral domain)—(moment method) procedure are interpreted phenomenologically through asymptotic reductions to yield truncation edge diffracted contributions, local Floquet modes from the overall bulk of the array, and individual scatterings from displaced elements. The contracting behavior of these contributions under high-frequency time-harmonic and short-pulse conditions is noted and interpreted. [Work supported by ONR.]

10:45

4SA8. Backscattered signature features in the combined time-frequency domain of submerged targets insonified by ideal sound impulses. Hans C. Strifors (Nat. Defense Res. Establishment, FOA 2, S-17290 Sundbyberg, Sweden) and G. C. Gaunaurd (Naval Surface Warfare Ctr., Res. Dept., R-42, Silver Spring, MD 20903-5000)

The backscattered pulses resulting when ideal acoustic impulses with plane wave fronts are incident on various submerged structures are studied. The structures are selected to be solid steel or granite spheres, and also spherical steel shells containing either air or water. A time-windowed Fourier transform (TWFT) and a pseudo-Wigner distribution (PWD) of the backscattered pulses are computed and displayed, in both cases employing a Gaussian time window. It is demonstrated that each of these distributions can extract and exhibit informative features that agree well with the general time development of the various kinds of resonances usually present in the considered frequency band (in this case: $0 < k_1 a < 60$) of these bodies. Bringing the returned signature information closer to its physical causes is a prime merit of those time-frequency distributions and of the displays they produce. Another one of their desirable properties is their ability to reduce the number of dominant signature features present in the echo while preserving the time progression of the extracted features. Both these properties will improve the target-recognition capability ordinarily furnished by the standard sonar cross sections of the considered targets.

11:00

4SA9. Scattering of impulse signal by elastic shells with internal structures. Yue-Ping Guo (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Sound scattering from an elastic shell can be significantly affected by structures loaded inside the shell. A major effect of internal loading has been shown to be the conversion of subsonic flexural waves in the shell into strongly radiating acoustic waves. In the frequency domain, this leads to a deeply scalloped scattering form function with the empty-shell result as its "mean value." Since flexural waves in the shell are much slower than supersonic compressional leaky waves that dominate the scattering process for an empty shell, the conversion of flexural waves into sound by internal structural loading is well separated in time from other processes such as the initial "bubble-like" reflection and the phase-matched leaky wave re-radiation. Also, because of the dispersive nature of the flexural waves, the internal loading effect in time domain can be clearly identified; they distinguish themselves in the scattered field by a series of wave packets whose amplitudes decay due to both acoustic damping and dispersion. This is in contrast to the other two main components in the scattered time signal, one being the initial reflection that is basically the inverse of the incident signal and the other being the leaky wave contribution consisting of decaying pulses. All these are clearly demonstrated in this paper, which examines the scattering of impulse incidence from elastic shells with internal structures.

The response of the loaded shell is also examined, which further reveals the interactions between the acoustic field, the shell, and its internals. [Work supported by ONR.]

11:15

4SA10. Radiated sound energy of a fluid-loaded cylinder as a function of force time duration and multiple pulse time spacing. Daniel H. Kruger and J. Adin Mann, III (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50011)

Transfer functions measured by near-field acoustical holography (NAH) are used to simulate the sound radiated by a point-driven, fluid-loaded cylindrical shell with spherical end caps in the time domain for a variety of force inputs [J. Acoust. Soc. Am. **90**, 1656–1664 (1991)]. The accuracy of time and frequency domain processing is reviewed. Radiated sound energy as a function of the time duration of a single pulse driving force is studied. The results are partially explained by looking at the output sound power as a function of frequency. Radiated sound energy as a function of the time spacing between multiple pulses is also studied. Sound radiation from the entire shell, the ends, and the driver alone are used to explain results. Frequencies determined by modal analysis as structurally resonant, acoustically radiating, both resonant and radiating, and neither resonant nor radiating were simulated for both single and multiple pulse forces. Instantaneous acoustic intensity, instantaneous power, and the time integral of these quantities are analyzed. Conclusions on the importance of time-domain analysis are made.

11:30

4SA11. Mid-frequency enhancement of the impulse response of a thin spherical shell: Ray approximation and a related frequency-domain analysis. Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A ray approximation was previously applied to the mid-frequency enhancement of the backscattering that occurs for ka close to the shell radius to thickness ratio a/h for thin shells [L. Zhang *et al.*, J. Acoust. Soc. Am. **90**, 2341(A) (1991) and L. G. Zhang *et al.*, J. Acoust. Soc. Am. **91**, 1862–1874 (1992)]. The enhancement is primarily associated

with the earliest guided ray contribution (that circumnavigates only the backside of the shell) of the slightly subsonic guided wave denoted by $l = a_{0-}$ by some authors. This ray contributes a large wave packet to the impulse response that is approximated in the present work. It is shown that the form function contribution $|f_{m=0,l}|$ is well approximated by a Gaussian centered on the ka where the damping $\beta_l \approx 1/\pi N_p/\text{rad}$. The resulting time-domain wave packet has an approximate Gaussian envelope and carrier frequency associated with that spectrum. The envelope amplitude is affected by the ka dependence of the phase velocity c_l . This ray analysis, although incomplete, also gives insight into the background-subtracted steady-state form function. [Work supported by ONR.]

11:45

4SA12. Ray acoustic formulation of sound wave scattering from submerged cylindrical thin elastic shells with endcaps. J. M. Ho and L. B. Felsen (Weber Res. Inst./Dept. of Elec. Eng., Polytech. Univ., Farmingdale, NY 11735)

Cylindrical shells with endcaps of various types are useful prototypes for a class of submerged elastic structures of current interest period. In the mid- and high-frequency ranges ($ka \geq 3$, where “ a ” is the radius of the cylinder), ray acoustics offers a physical-based methodology for computing the scattered signature. Restricted to thin shell dynamics, hemispherically capped cylinders are investigated by combining results from our previous studies of infinite cylindrical and full spherical shells [Felsen *et al.*, J. Acoust. Soc. Am. **87**, 554–569 (1990); J. M. Ho and L. B. Felsen, J. Acoust. Soc. Am. **88**, 2389–2414 (1990)]. Rays from arbitrarily incident plane waves are traced to the cylinder and give rise there to geometrically reflected and phase matched shell guided supersonic wavefields, each with its own distinct trajectories. The surface guided fields follow helical paths on the cylindrical portion, which are smoothly matched to great circle paths on the hemispheres, and are turned back there toward the cylinder at a pitch-dependent caustic. The total field is synthesized by the sum of all possible ray paths, traversing the structure azimuthally and longitudinally, which reach the observer. Quantitative expressions are obtained by including ray field launching, detachment, and propagation coefficients. Attention is also given to conical endcaps and to conical sections with smoothly joined spherical caps. [Work supported by ONR.]

WEDNESDAY MORNING, 13 MAY 1992

SALON E, 8:20 A.M. TO 12:25 P.M.

Session 4SP

Speech Communication: Acoustic and Articulatory Phonetics of Dialects

Alice Faber, Cochair

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

Marianna DiPaolo, Cochair

Linguistics Program, Stewart Building, University of Utah, Salt Lake City, Utah 84112

Chair's Introduction—8:20

Invited Papers

8:25

4SP1. Types of dialect variation. Malcah Yaeger-Dror (Linguistics, Univ. of Arizona, Tucson, AZ 85721)

Speech variation is of importance to workers in automated speech recognition and synthesis. This paper will discuss specific sources of variability that are of practical concern to these researchers. Quantitative

evidence will be presented from American dialects of both English and French to demonstrate how vowel formant frequency variation is correlated with both the sonority of intrasyllabic following consonants and with prosodic information. Evidence will also be presented from analysis of vowel formant information from various speech "styles" and "registers" to demonstrate how researchers can appropriately integrate such variation into synthesis modules. Evidence for several sources of variation will be presented, and both theoretical and practical conclusions will be drawn from the evidence. [Work partially supported by NSF.]

8:50

4SP2. Controlling for dialect in contextless vowel identification: Revisiting the Peterson-Barney experiment. Sharon Ash (Dept. of Linguistics, 619 Williams Hall, Univ. of Pennsylvania, Philadelphia, PA 19104-6305)

In an early study on the spectrographic measurement of vowels, Peterson and Barney proposed that listeners would interpret a vowel heard in isolation as the phoneme in their own speech corresponding to that sound [G. E. Peterson and H. L. Barney, *J. Acoust. Soc. Am.* 24, 175-184 (1952)]. The present study is based on their experiment and aims to test their hypothesis by controlling for the dialect of both speakers and listeners. In this investigation, listeners in Philadelphia, Birmingham, and Chicago were asked to recognize 14 vowel phonemes pronounced in the frame /k__d/ by speakers from the same three cities. It is found that for most firmly established variables, local listeners have an advantage in identifying phonemes pronounced by speakers of their own dialect. However, for new changes in progress and for at least one well-known stable variable, local listeners have great difficulty identifying the vowel, nearly as much as do listeners from other areas. Finally, both local and nonlocal listeners make a significant number of errors in identifying certain phonemes. The findings are explained in terms of chain shifts and the phonetics of the test tokens. [Work supported by NSF.]

9:15

4SP3. The inhibition of speech perception by linguistic categories. William Labov (Dept. of Linguistics, Univ. of Pennsylvania, 1105 Blockley Hall, Philadelphia, PA 19101)

In a number of empirical studies it is reported that speakers consistently make a distinction between certain word classes, but categorize them as "the same" in perception tests. This paper reports experiments that explore the ability of Philadelphians to categorize and discriminate /er/-/ɛr/ in *ferry*, *furry*, etc. A first series of categorization experiments showed that Philadelphians are significantly worse than non-Philadelphians in the ability to label and interpret this distinction. A second experiment combined tests of categorization and discrimination. Stimuli separated by 100-Hz differences in F_2 were used to construct pairs in a 4IAX model, where subjects judged which of two pairs were different. Non-Philadelphians showed a linear psychophysical response, from near 100% for a three-step discrimination to 78% correct for a one-step discrimination. Philadelphians are much inferior in discrimination for the three-step test, but the difference disappears as discrimination becomes more difficult in the one-step test. The linguistic decision that two sounds are "the same" appears to interfere with discrimination to the extent that the task is recognized as a linguistic one. [Work supported by NSF.]

9:40

4SP4. Acoustic and auditory evidence of sociophonetic variation in Vancouver: LTAS and vocalic data. John H. Esling (Dept. of Linguistics, Univ. of Victoria, Victoria, BC V8W 3P4, Canada)

Formant frequencies of ten Canadian English vowels from identical environments in reading style were calculated for 128 subjects in the middle and older age groups of the sociolinguistic Survey of Vancouver English. LPC-based F_1 , F_2 distributions were derived using a computerized speech lab (CSL) program and evaluated statistically to identify differences in vowel quality across four socioeconomic-status (SES) classes for both sexes. Long-term average spectra (LTAS) of 60 s of reading text were also computed and compared with vowel formant results. The SDDD LTAS dissimilarity index (standard deviation of the differences distribution) was also applied to the 192 subjects in all (including young) age groups. Auditory evaluations of each subject's production of each vowel were compared with the results of formant analysis and both LTAS analyses. Here, F_1 values were more uniformly differentiated than F_2 , and some groups' vowels were more vulnerable to rejection of F_2 on bandwidth criteria than others. Differentiation is significant and consistent for a high proportion of vowel formant distributions for some SES groups, notably between

working-class (WC) and middle-class (MC) middle-aged women. Some F_1 , F_2 patterns are compatible with auditory predictions of a hierarchy of vocalic susceptibility to contrasting voice quality settings. [Work supported by SSHRCC.]

10:05

4SP5. Acoustic variability in the vowels of female and male speakers. Caroline Henton (Apple Computer, Inc., 20525 Mariani Ave., MS76-7E, Cupertino, CA 95014)

When vowel formant values for female and male speakers are normalized and plotted in an F_1 - F_2 space, it is striking that the quadrilateral spaces for the females are uniformly larger than those for the males. Data will be presented from seven languages and dialects, including three dialects of English, which indicate that, all other things being equal, female speakers appear to produce vowels in a manner that is more phonetically explicit than males do. It is particularly in the F_1 dimension that the females' quadrilaterals extend beyond the males'; it may be inferred that female speakers articulate vowels with a more open mouth than males do. Can one expect that speakers with a higher F_0 automatically have a larger vowel space? Are females making more of an effort to keep vowels distinct, and potentially contribute to greater intelligibility? Or do females over-articulate, avoiding reduced or centralized forms, as a result of social expectations to be conservative "guardians" of the language? Acoustic and sociophonetic answers to such questions will be offered. In addition, data will be presented from three dialects of English which show that female speakers are not uniform in their behavior: some females merge vowels more often than males do, while other female populations appear to differentiate the same vowels more systematically. The perceived social prestige of an accent is offered as one explanation for these disparate directions of change.

10:30

4SP6. Durational effects of prosodic structure in spontaneous spoken French. Henrietta J. Cedergren, Louise Levac, and Hélène Perreault (Dept. de Linguistique, Univ. du Québec à Montréal, C.P. 8888, Montréal, Quebec H3C 3P8, Canada)

The relation between segmental timing variability and prosodic structure in Montreal French is investigated. Twenty-four minutes of data extracted from recordings of spontaneous conversations using a balanced corpus of eight speakers differentiated according to sex, age, and social class were measured on digital spectrograms. An auditory analysis was used to categorize the basic prosodic structure of the speech corpora. Four levels of prosodic organization were established: syllabic units, metrical feet, accentual groups, and intonational phrases. A linguistic analysis was used to determine the distribution of deleted vowels. This study investigates the role of two types of durational effects in utterances: local effects and global effects. The multiple layering of temporal effects is examined by means of a statistical analysis of syllable durational variability. It is shown how the distribution of vowel deletion follows from the interaction of effects in the temporal implementation of prosodic domains. [Work supported by SSHRCC.]

10:55-11:10

Break

Contributed Papers

11:10

4SP7. Using the method of adjustment to study vowel spaces. Keith Johnson, Richard Wright, and Edward Flemming (Dept. of Linguistics, UCLA, 405 Hilgard Ave., Los Angeles, CA 90024-1543)

Comparing the acoustic vowel spaces of different dialects or languages is fraught with peril because acoustic measurements confound personal and linguistic information. The study to be reported addressed this problem by using the method of adjustment. Listeners were asked to adjust the F_1 and F_2 of synthetic vowels to produce a perceptual match between the synthetic vowel and the vowel sound they would

expect to hear in a visually presented word. The monosyllabic test words represented ten or eleven different vowel qualities (depending on dialect) for American English speakers. The listeners adjusted F_1 and F_2 by moving a mouse cursor, clicking the mouse button to hear the vowel sound indicated by the location of the cursor on a CRT. F_3 , F_4 , and the formant bandwidths were estimated by regression from previously reported acoustic data. Preliminary results suggested that: (1) Speakers of the same dialect were very consistent with each other; within subject variability (which was small) accounted for about 70% of all within vowel variability. (2) The method revealed perceptual vowel space differences between speakers of different dialects of American English. (3) Differences between monolingual and bilingual speak-

11:25

4SP8. VOT in Norwegian stops. Berit Halvorsen (Dept. of Linguistics and Phonetics, Univ. of Bergen, Bergen, Norway and Haskins Laboratories, New Haven, CT 06511)

The stops from three Norwegian dialect areas [Eastern Norwegian (EN), Trønder dialects (TD), and the Bergen dialect] were recorded. Native speakers of Norwegian read mono- and disyllabic words twice in a carrier phrase. Stressed vowels were [e:], [ɛ], [u:], and [o]. All three dialects have /bdgptk/, EN and TD have retroflex stops /tɖ/ and TD also have palatal stops /cɣ/. Measurements of VOT were made using a wave-editing program at Haskins Laboratories. It was hypothesized that (1) the traditionally labeled "voiced" stops would have zero or short lag onset. This turned out to be the case for most of the speakers, but some of them had extraordinarily long lead onsets. Free speech from the same subjects showed no or smaller lead onsets in /bdg/, however. (2) VOT in labials < dentals/alveolars < velars. Measurements showed clear distinction in VOT between labial and dental/alveolar stops, but there was smaller difference in VOT for dentals/alveolars and velars. These results will be corroborated with perception tests with synthesized stops in CV syllables. [Work supported by the Norwegian Research Council for the Humanities.]

11:40

4SP9. Register and speech variation. Malcah Yaeger-Dror (Dept. of Linguistics, Univ. of Arizona, Tucson, AZ 85721) and Jay Nunamaker (MIS Dept., Univ. of Arizona, Tucson, AZ 85721)

There are several speech parameters that are known to vary radically with different speech registers. One of the primary concerns of sociolinguists has always been the degree to which speech "style" and "register" influence speech production; this variability is now of greater concern to members of Acoustical Society because the variation influences intelligibility of speech for machines, and of synthesized speech. This paper will discuss evidence that pitch contours, vowel reduction, and vocabulary choice are all influenced by register. This paper will focus on the use of the work "not," analyzing the relative likelihood of pitch prominence, neutral presentation, or reduction (*n't*) in three different registers. Data from political debates will be compared with data from small group "brainstorming" sessions and with data from a DARPA workshop. There will be a discussion of the importance of this variation to both automatic speech recognition and speech synthesis. [Work partially supported by NSF.]

4SP10. /l/ and /ng/ effects on vowel nuclei in four dialects. Thomas Veatch (Dept. of Linguistics, Stanford Univ., Stanford, CA 94305-2150)

The effects of following /l/ and /ng/ on vowel quality as measured by nucleus F1 and F2 frequencies are compared across four vernacular dialects, including Alabama White, Chicago White, Los Angeles Chicano, and Jamaican Creole. Effects of both following /l/ and following /ng/ on vowel nuclei vary across dialects. The phonetic implementation rule by which /l/ is realized as darker or lighter [Sproat and Fujimura (1990)] is absent in Jamaican Creole, which has a lighter /l/ and different effects (/l/ only retracts back, non-low vowels). Some anomalous effects are given either phonological or phonetic explanations. /ng/ raises preceding /l/ consistently in all four dialects except Alabama, where /ng/ lowers the nuclei of preceding vowels including /l/. Since /ng/ presumably does not differ from one dialect to another, this difference cannot be mechanical. Nor is it a phonological effect, it is argued. Thus two strictly phonetic patterns are shown to be present in some dialects and absent or different in others. This is additional evidence for a linguistic component of the phonetic system by which phonological structure is realized as sound.

12:10

4SP11. Vowel reduction and perceptual recovery in casual speech. Sharon Y. Manuel (Dept. of Commun. Disord. and Sci., Wayne State Univ., 563 Manoogian, Detroit, MI 48202 and Res. Lab. of Electron., MIT, Cambridge, MA 02139)

A pervasive characteristic of casual (normal conversation) English is the apparent deletion of unstressed vowels like the first vowel in the word "support." One might suppose that if the first vowel in "support" were deleted, "support" would become homophonous with "sport." Acoustic and physiological data are reported which suggest that in fact when speakers appear to be deleting an unstressed schwa, they are often actually omitting only the oral gestures for the vowel. The glottal gestures stay much as they are in careful speech and are tied to the remaining oral gestures much as they are in careful speech. Also reported are perceptual data which suggest that listeners are sensitive to the acoustic consequences of these residual patterns, and can use this information to distinguish between "sport" and reduced versions of "support" [see also J. Fokes and Z. S. Bond, *J. Acoust. Soc. Am. Suppl.* 1 85, S85 (1989); J. Fokes and Z. Bond, *Proc. XII Int. Congr. Phon. Sci.* 4, 58-61 (1991)]. These results will be discussed in terms of the more general issue of deletion and recovery of phonetic information. [Work supported by NSF.]

Session 4UW

Underwater Acoustics: Propagation I

Stewart A. L. Glegg, Chair

Department of Ocean Engineering, Florida Atlantic University, Boca Raton, Florida 33431

Chair's Introduction—7:55

Contributed Papers

8:00

4UW1. A review of the Parabolic Equation Workshop II. David B. King, Stanley A. Chin-Bing (Naval Res. Lab., Stennis Space Center, MS 39529), James A. Davis (Planning Systems, Inc., New London, CT 06320), and Richard B. Evans (Sci. Applications Intl. Corp., New London, CT 06320)

In Spring, 1981, the first Parabolic Equation (PE) Workshop was held [Davis, White, and Cavanagh, NORDA Technical Note 143 (1982)]. The purpose of that workshop was to provide a forum for theoretical and applied PE developers and to compare computer results for a set of ocean acoustic problems. Underwater acoustic modelers have benefited from that set of test cases and documented PE deficiencies for nearly a decade. The second PE workshop was recently held (in Spring, 1991) to document the impressive advances that have taken place in the development of PE models over that decade. PE developers from the ocean acoustics community were invited to provide computer solutions for a new set of ocean acoustic problems. The set of problems was designed to test the ability of PE models to handle (a) wide-angle propagation (nearly $\pm 90^\circ$ with respect to the axis of propagation), (b) shear waves and shear attenuations, (c) backscatter, (d) conservation of energy, and (e) complicated range-dependent environments. In addition, predictions from the PE models were compared with measured acoustic data taken in a region where the ocean environments (water column, sediment, ocean bottom, and subbottom) were well-known. Results from this PE Workshop II indicate that several significant advances have occurred in PE model development to the extent that some current PE models can include very wide-angle propagation, shear waves, and backscatter, while conserving energy. Additionally, an unexpected problem experienced by some wide-angle split-step PE models was also presented at the PE Workshop II. These and other results from the PE Workshop II will be presented and discussed. [Work supported by ONR AEAS and ONR Acoustic Reverberation SRP.]

8:15

4UW2. Computational comparison of broadband acoustic models. Michael H. Brill and Anthony I. Eller (Sci. Applications Intl. Corp., 1710 Goodridge Dr., McLean, VA 22102)

Broadband acoustic propagation models vary greatly in their computation times. To define the speed-versus-accuracy tradeoffs, two questions are addressed: (1) For ducted environments, how does the para-

bolic equation (PE) model compare with the faster Labianca virtual-mode model (VM) and the FAME eigenray part of the generic sonar model (GSM)? (2) In nonducted environments, must a GSM complex transmission loss (TL) be computed for all the signal frequencies, or is a single frequency of computation sufficient (i.e., is it fair to assume that the only frequency dependence in the complex-TL phase is in the ray time delay)? To answer these questions, the effects of the computed complex-TL on sine-wave-burst and HFM-burst signals are presented. The multipath arrival structures of the environments enforced a 4-s received-signal duration, and the transmit signals enforced a bandwidth of 200 Hz. The results are as follows: (1) A weak duct shows favorable comparison between GSM, VM, and PE models; however, a strong duct does not, and in fact reveals numerical instability in the PE computations. (2) Single-frequency GSM captures the basic arrival structure of multiple-frequency GSM, but the multipath interference is different and sensitive to the dominant signal frequency. It is concluded that GSM (but not single-frequency GSM) may in most practical cases yield adequate predictions for the acoustic filtering of broadband signals.

8:30

4UW3. Comparison of numerical predictions for transmission loss and reverberation levels in shallow waters. G. Gray and G. C. Gaunard (Naval Surface Warfare Ctr., White Oak, Silver Spring, MD 20903-5000)

Since there are no models to currently predict transmission losses (TL) and reverberation levels (RL) for sound propagation problems in shallow waters, what is customarily done to estimate these parameters is to use various standard numerical codes in the same form as they were originally developed for deep waters. This introduces a variety of errors that one must live with, until suitable models are specifically developed for the various shallow water environments of interest. In order to begin to understand the nature and size of these errors (which could often be larger than 60 dB, i.e., that is a factor greater than one million!), half-a-dozen commonly used TL and RL prediction codes have been considered and have been applied to six geographical sites and to fourteen (conceptual but realistic) configurational cases. The models used are the generic sonar model (GSM), the BAM, COLOSSUS, NISSM, RAYMODE, RASP, and RUMBLE. An extensive comparison between the predictions of all these codes in all the cases and sites was carried out, and the results were compared to the (sparse) available measured

data. An N.S.W.C. report with the details is in preparation. The study reveals that COLOSSUS provides the "best" TL predictions (i.e., closest to the data) and RASP the best ones for RL, in the shallow water environments found in the continental shelves.

8:45

4UW4. PE modeling of upslope propagation data. David J. Thomson, Garry J. Heard, and Gary H. Brooke (Defence Res. Establishment Pacific, FMO Victoria, BC V0S 1B0, Canada)

The analysis and interpretation of underwater acoustic propagation data measured in regions of complex bathymetry (e.g., near continental slopes) can be enhanced by using a full-wave, range-dependent propagation model. Such models are particularly important at low frequencies, where sub-bottom penetration of sound can be significant. With the recent development of a two-way parabolic equation (PE) formalism [e.g., M. D. Collins *et al.*, *J. Acoust. Soc. Am.* **90**, 2277 (1991)], fast and accurate modeling of forward- and backscattered waves in range-dependent media can be realized. For propagation in an idealized wedge-shaped waveguide, the outgoing field computed with this two-way PE model compares well with the total field given by a more computationally intensive, coupled-mode model. In this paper, the formulation of a two-way PE model is used to interpret shot data recorded during an upslope experiment off the west coast of Canada. The shot waveforms were received on a 1200-m towed line array that opened range along a 15-km track. Over this endfire propagation run, the water depth decreased from 1500 to 500 m. The measured data were processed as a function of frequency, range, and vertical angle and compared to the forward-scattered component of the two-way PE field.

9:00

4UW5. Abstract withdrawn.

9:15

4UW6. Wave field splitting, invariant imbedding, and phase space analysis applied to ocean acoustic wave propagation modeling. Louis Fishman (Dept. of Math. and Comput. Sci., Colorado School of

Mines, Golden, CO 80401)

Due to the extremely large size of realistic three-dimensional ocean problems, there is an understandable desire to incorporate marching methods into the solution algorithm for the inherently elliptic (frequency-domain) problem. Recognizing that typical ocean propagation problems are essentially scattering problems in terms of a transition region and transversely inhomogeneous asymptotic half-spaces, wave field splitting, invariant imbedding, and phase space methods reformulate the problem in terms of an operator scattering matrix characteristic of the transition region. The subsequent equations for the reflection and transmission operators are first order in range, nonlinear (Riccati-like), and, in general, nonlocal. The system is well-posed, but stiff. The reflected and transmitted wave fields can be computed in a very efficient manner, while the wave field in the transition region (if desired) can be computed by essentially a layer-stripping algorithm. In principle, the transition region can be divided into subregions, allowing for parallel computations and subsequent recombination. Numerical examples will be discussed. [Work supported by NSF, AFOSR, ONR, ASEE, JEWIC.]

9:30

4UW7. Massively parallel simulation of the ideal wedge benchmark and variations. R. A. Krutar, S. K. Numrich, and J. B. Cole (U.S. Naval Res. Lab., Code 5126, Washington, DC 20375-5000)

An efficient lattice gas scheme for simulating the propagation and scattering of underwater sound over the entire near- to far-field range has been implemented on a Connection Machine at NRL. This method has been tested on the ideal wedge benchmark. At 2-m resolution 8000 time steps were calculated and 381 frames consisting of 512×512 cells each were rendered (a total of 2 billion cell updates) in less than 10 min. The utility of this simulation technique is illustrated by introducing variations into the basic problem, that would be difficult to model using normal mode theory.

9:45

4UW8. Parallel and vector algorithms for solving the underwater parabolic equation with a finite element model. R. A. Zingarelli (Naval Res. Lab., Acoustic Theory and Simulation Code 221, Stennis Space Center, MS 39529)

Rapid solution of the parabolic equation via the finite element method in a range-dependent environment is necessary for practical applications. Several backward recurrence relations inherent in the solution inhibit efficient utilization of specialized computer hardware. Three algorithms suitable for vector, parallel, and massively parallel hardware are presented. Efficiency, speed, numerical stability, and application to existing problems are discussed. [Work supported by ONT.]

10:00-10:15

Break

10:15

4UW9. A three-dimensional analysis of acoustic propagation in a penetrable wedge. Grant B. Deane (Mail Code 0238, Marine Phys. Lab., Scripps Inst. of Oceanography, La Jolla, CA 92093)

The method of source images is used to develop an analytical solution for the three-dimensional acoustic field in a penetrable wedge. The acoustic field for a harmonic, point source is computed assuming isovelocity sound field profiles in the water and sediment. The solution for the field consists of a sum of contributions due to source images and

gives the sound field at any point in the water or sediment of the wedge. This analysis differs from the earlier work of Deane and Tindle (submitted to JASA, 10 December 1991), which is restricted to propagation in a plane perpendicular to the wedge apex. The present work uses a Bessel function expansion that results in a computationally efficient solution to the field which is valid throughout the wedge.

10:30

4UW10. Recursive ray acoustics for three-dimensional sound-speed profiles. Lawrence J. Ziomek (Dept. of Elec. and Comput. Eng., Code EC/Zm, Naval Postgraduate School, Monterey, CA 93943)

A recursive ray acoustics (RRA) algorithm for three-dimensional sound-speed profiles is presented. The RRA algorithm is simple, fast, and accurate and uses arc length as the independent variable. It can be used to compute the position, angles of propagation, travel time, and path length along a ray path. The accuracy of the RRA algorithm was tested by comparing its results with those obtained from a ray acoustics algorithm that requires the solution of a system of four, first-order, ordinary differential equations (ODEs). The ODE algorithm, which uses horizontal range as the independent variable, can only handle one-dimensional, depth-dependent, sound-speed profiles (SSPs). Therefore, five standard, depth-dependent, SSPs were used to test the two algorithms. The results from the ODE algorithm were treated as the benchmark, with respect to accuracy, for comparison purposes. The RRA algorithm proved to be very accurate for the test cases tried. After the accuracy of the RRA algorithm was validated by using one-dimensional SSPs, its three-dimensional sound-speed capability was also tested. [Work supported by DARPA and the Naval Postgraduate School Direct Funded Research Program, sponsored by ONR, Code 1125 OA.]

10:45

4UW11. Acoustic modeling through a Mediterranean water lens (Meddy). David G. Browning, George Botseas, and Eugene M. Podeszwa (New London Detachment, NUWC—Newport Div., New London, CT 06320)

Mediterranean water lenses (Meddies), as reported by Armi [L. Armi and W. Zenk, *J. Phys. Oceanogr.* **14**, 1560–1576 (1984)], are small warm-core eddies with a large width (100 km) to thickness (0.9 km) ratio. Centered at a depth of approximately 1000 m, they were observed in the Canary Basin of the North Atlantic Ocean. These are unlike many warm-core eddies that extend to or near the surface [P. A. Nysen, P. Scully-Power, and D. G. Browning, *J. Acoust. Soc. Am.* **63**, 1381–1388 (1978)], and hence may cause unique propagation anomalies for certain source–receiver configurations. A set of reported oceanographic specifications of a Meddy was converted to sound-speed profiles which were merged with historical data below a depth of 2000 m.

Acoustic modeling runs at 200 Hz were made across the Meddy using the IFD/PE model. The results were compared to normal propagation predictions for the area. The characteristic shift of the convergence zones is observed, but as might be expected the impact is relatively small for shallow source–receiver configurations and greatest for source and receiver depths near the center of the Meddy.

11:00

4UW12. Modal travel time and dispersion with an eddy lens (EL). E. C. Shang and Y. Y. Wang (CIRES, Univ. of Colorado/NOAA/Wave Propagation Lab., 325 Broadway, Boulder, CO 80303)

A special type of mesoscale eddy, called intrathermocline eddy or eddy lens (EL), has been studied intensively in oceanography over the past decades. The special features of the EL are (1) the deep warm core (near the sound channel axis), (2) the strong anomalies of sound speed (can be as large as 10–15 m/s), and (3) the long lifetime. The significant impact on acoustic wave propagation of EL due to these features is the “double channel” effect. In this paper, mode couplings caused by the “double channel” induced by EL is investigated. For ocean acoustic tomographic interest, modal travel time and modal dispersion are calculated by using the modal spectrum of the PE field (MOSPEF) method. Numerical simulation for a 57-Hz pulse with 10-Hz bandwidth is conducted. [Work supported by NOAA.]

11:15

4UW13. On equations for the speed of sound in seawater. Brian D. Dushaw, Peter F. Worcester, Bruce D. Cornuelle (Scripps Inst. of Oceanography, Univ. of California, San Diego, La Jolla, CA 92093-0208), and Bruce M. Howe (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Long-range acoustic transmissions made in conjunction with extensive environmental measurements and accurate mooring position determinations have been used to test the accuracy of equations used to calculate sound speed from pressure, temperature, and salinity. The sound-speed field computed using the Del Grosso equation [V. A. Del Grosso, *J. Acoust. Soc. Am.* **56**, 1084–1091 (1974)] give predictions of acoustic arrival patterns which agree significantly better with the long-range measurements than those computed using the Chen and Millero equation [C. Chen and F. J. Millero, *J. Acoust. Soc. Am.* **62**, 1129–1135 (1977)]. The predicted ray travel times and travel time error have been calculated using objectively mapped sound-speed fields computed from CTD and XBT data. Using the measured and predicted ray travel times, a negligible correction to Del Grosso's equation of 0.05 ± 0.05 m/s at 4000-m depth is calculated. Small errors of about 50 m in the GPS determination of mooring positions lends a depth-independent error of 0.1 m/s to the sound-speed equation correction. [Work supported by NSF and ONR.]

Meeting of Accredited Standards Committee S2 on Mechanical Shock and Vibration

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
4615 O'Meara Drive, Houston, Texas 77035

Standards Committee S2 on Mechanical Shock and Vibration. 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, to take place in London, United Kingdom, from 22 March–2 April 1993.

Session 5EA

Engineering Acoustics: Pioneering in Acoustics

Robert D. Finch, Chair

Department of Mechanical Engineering, University of Houston, Cullen College of Engineering, Houston, Texas 77204-4792

Chair's Introduction—1:00

Invited Papers

1:05

5EA1. Architectural acoustics: Some pioneering and some prognostication. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138)

Architectural acoustics, following Wallace Sabine in 1900, entered a new era after World War II. Measuring equipment was improved—new sound level meters, microphone calibrators, graphic level recorders, octave-band analyzers, and, soon, magnetic tape recorders. The National Bureau of Standards, the Riverbank Laboratories, and others had armed us with data on the absorption of sound by commercial acoustical materials and the transmission loss of standard walls. The broadcasting industry taught studio design with thin molded plywood surfaces. At the risk of delimiting the broader scope of the field, including noise control in building spaces, this paper reviews the very demanding area of room design for listening and performance of unamplified concert music, choral works, and opera. Recent studies have shown that an extraordinarily delicate balance exists among the many interrelated variables that comprise an optimum environment for music. The attributes of acoustics that must be satisfied in design, and several alternate architectural shapes of noteworthy recent halls are reviewed. An exciting new program, just underway by an international consortium of researchers and consultants, to systematically measure and document existing halls throughout the world is described.

1:30

5EA2. Progress in environmental noise control. W. J. Cavanaugh (Cavanaugh Tocci Associates, Inc., 327F Boston Post Rd., Sudbury, MA 01776)

Environmental noise, i.e., unwanted sound around and within the buildings in which people work and live, is an ubiquitous byproduct of mankind's societal and technological development. Noise exposure of populations throughout the developing world gradually accelerated starting with the industrial revolution of the 19th century and increased dramatically with the population growth and introduction of new transportation and other sources during the 20th. In recent decades, people who sought the tranquility of suburban living were increasingly greeted with new industrial parks whose tenants also sought escape from decaying, no longer economically sound, urban locations. Even the return to regenerating city centers assures little relief from the noise of new highway, rail, and expanding airport developments or, for that matter, from noisy air conditioning equipment serving buildings nearby or from the sounds of one's own building neighbors above, below, or next door. This paper reviews the last 40 or so years of what must be considered remarkable progress in dealing with the diverse challenges of environmental noise. In terms of the classical "systems" approach to noise control, the sources and transmission paths are now reasonably well developed and understood both in theory and in measurement. The effects of environmental sounds that ultimately reach the human receptors are less so, however. The continuing development of adequate criteria represents the significant remaining challenge in environmental noise control. This paper reviews some of the pioneering criteria developments for background noise and acoustical privacy in buildings and for community noise.

1:55

5EA3. Acoustical measurements and instrumentation. Gunnar Rasmussen (Brüel & Kjær, 2850 Nærum, Denmark)

Acoustical measurements are fully accepted as precise and well-defined techniques for establishment of noise and sound quality criteria. A long line of national standards has enabled the creation of a large database with comparable results supporting confidence in the field of acoustics. The half-inch microphone is now the industry standard minimizing errors related to operator ignorance. The use of correctly measured rms values is taken for granted due to progress in instrumentation detectors. The acousticians have available

well-standardized filters for frequency analysis. Intensity measuring techniques are now available along with calibration techniques offering the acoustician the possibility to contribute to the field of structural vibration, machine diagnostics, product development, quality control on machines and products in general, condition monitoring, etc. Acoustic transducers and techniques are used for gas analysis, medical diagnostics, and many other fields. The acoustician who understands physics, dynamics, and wave propagation has a great chance to influence the technical process. Frequency analysis of steady continuous signals has been taught very well in the educational sector along with finite element modeling in the context of digital processing on computer-based systems. The far more important treatment of real world dynamic and often nonlinear and distributed processes represents the future challenge to the engineer and therefore to the researcher and educator.

2:20

5EA4. Pioneering in acoustics: Hearing aid transducers. Mead C. Killion (Etymotic Res., Elk Grove Village, IL 60007 and Northwestern Univ., Evanston, IL 60208)

Three men, Sam Lybarger, Hugh Knowles, and Elmer Carlson, have been responsible for a disproportionate share of the hearing-aid developments over the last 60 yr. Their technical contributions, and the importance of those contributions to better hearing aids, will be reviewed. Two of them are still healthy and productive, indicating that making the world better for the hearing impaired may slow the aging process.

2:45

5EA5. Underwater transducers. Stanley L. Ehrlich (Stan Ehrlich Associates, P.O. Box 3679, Newport, RI 02840)

The set of underwater acoustic transducers of primary interest in this presentation is employed to provide electrical signals to sonar receivers, to accept electrical outputs from sonar transmitters, or both. Several different transducer element and array configurations will be selected for discussion based on the author's personal experience and the open literature in reports and journals, including review papers and patents. A brief overview of the active materials and mechanisms used and the need for further improvement will be noted. Also, the impact of system synthesis versus the simpler separate element design will be discussed, since this is a major problem in "big" sonars. Finally, an attempt will be made to estimate the future trends in research and development of underwater transducers.

3:10-3:40

Panel Discussion

PANEL MODERATOR: Robert D. Finch

PANEL MEMBERS: Leo L. Beranek

William J. Cavanaugh

Stanley L. Ehrlich

Mead C. Killion

Gunnar Rasmussen

Session 5MU

Musical Acoustics: Distinguished Lecture: Acoustics and Orchestral Sound

Uwe J. Hansen, Chair

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

Chair's Introduction—3:10

Invited Paper

3:15

5MU1. Acoustics and orchestral sound. Jürgen Meyer (Physikalisch-Technische Bundesanstalt, Bundesallee 100, Postfach 33 45, D-3300 Braunschweig, Germany)

The tone quality of an orchestra rests on the cooperation of a multitude of differing instruments which are used partly in solo fashion and partly gathered in groups; it is furthermore shaped by the spatial extension of the ensemble and by the influence of room acoustics. Acoustic measurement procedures permit an analysis and description of tonal characteristics of individual instruments in regards to sound power output, spectral composition, time-dependent fine structure, and orientation dependence of sound radiation. Appreciation of the full orchestral sound, on the other hand, requires consideration of sound transit times within the orchestra with the associated problem of playing together, as well as variations in wall reflection times for different instrument positions. The effect of dynamics and masking, the chorus effect and the seating arrangement of the strings, and spectral and time-dependent tone construction will be demonstrated with the assistance of the Weber State University Symphony Orchestra, prepared by its regular conductor, Professor Michael Palumbo. Sound examples are taken largely from the overture to the opera "Der Freischütz" by Carl Maria von Weber. In conclusion, the orchestra will perform the entire overture directed by Professor Meyer.

Session 5PA

Physical Acoustics: Thermoacoustics, Drops, and Bubbles

Steven L. Garrett, Chair

Department of Physics, Code PH/Gx, Naval Postgraduate School, Monterey, California 93943

Chair's Introduction—1:25

Contributed Papers

1:30

5PA1. Standing wave analysis of the low amplitude performance of a thermoacoustic prime mover. Anthony A. Atchley (Physics Dept., Naval Postgraduate School, Monterey, CA 93943)

A standing wave analysis of the low-amplitude performance of a thermoacoustic prime mover is presented. The analysis, motivated by Swift [G. W. Swift, *J. Acoust. Soc. Am.* **84**, 1145–1180 (1988)], assumes that the acoustic pressure and velocity can be described in terms of a standing wave throughout the prime mover. The analysis is not limited by either the small boundary layer or short stack approximations. The results of the analysis are compared to measurements of the quality factor Q of a prime mover both below and above onset. Measurements were made in both a helium- and an argon-filled prime mover for mean pressures ranging from approximately 80 to 500 kPa. Below onset the Q is determined from the frequency response of the prime mover. Above onset the Q is determined from the initial rise time of the acoustic oscillations. In general, the agreement is good, indicating that previous work on finite amplitude standing waves [A. B. Coppens and J. V. Sanders, *J. Acoust. Soc. Am.* **58**, 1133–1140 (1975)] can, perhaps, be used to explain the observed high-amplitude waveforms in prime movers. [Work supported by ONR and the NPS Res. Prog.]

1:45

5PA2. Theoretical study of heat transport in thermoacoustic stacks having arbitrary plate separation. Thomas J. Hofter (Phys. Dept., Naval Postgraduate School, Code PH/HF, Monterey, CA 93943)

Thermoacoustic heat engines can be simply described as devices where heat transport is caused by the interaction between a standing wave and a stationary, parallel plate structure called the stack. Simplified calculations of this heat transport have been discussed previously, which assume perfect standing waves (infinite standing wave ratio) and plate separations that are large relative to a thermal penetration depth. A theory will be presented that is generalized to describe systems with arbitrary plate separation and SWR. Using this theory, the spectrum of heat engine designs that span "classic" thermoacoustic engines on the one extreme, and Stirling cycle engines on the other extreme, can be compared. Some results are that Stirling cycle engines can be more efficient than standard thermoacoustic engines and that, given some practical constraints, thermoacoustic engines can deliver more power than a comparable Stirling engine. Other results may help to design an improved engine that combines the characteristics of both thermoacoustic and Stirling engines. [Work supported by ONR and NRL.]

2:00

5PA3. Impedance and stability analysis of two thermoacoustic prime movers. James R. Belcher, W. Pat Arnott, Henry E. Bass, and Richard Raspet (Natl. Ctr. for Phys. Acoust. and the Dept. of Phys. and Astron., Coliseum Dr., University, MA 38677)

Thermoacoustic heat engines can be used to produce sound from heat and to transport heat using sound. The air-filled prime mover studied is a quarter wavelength resonator that produces sound at nominally 116 Hz for a temperature difference of $\Delta T = 176$ K. Specific acoustic impedance at the mouth of the prime mover was measured as a function of the temperature difference between the hot and cold heat exchangers. The real part of the impedance changes sign for sufficiently large temperature differences, indicating the possibility of sound production. The theoretical expression for radiation impedance of an open pipe was overlaid on the impedance curves. The operating point was confirmed from the intersection of experimental and theoretical impedance curves. The stability curve was computed for a helium-filled prime mover as a function of the ambient pressure, P_0 , for the first two modes. The first mode has a minimum $\Delta T = 154.6$ K for $P_0 = 173$ kPa and the second mode for $\Delta T = 455.4$ K for $P_0 = 200$ kPa. The resonant frequencies of these modes are approximately 310 and 610 Hz. The quality factor, which characterizes the exponential decay or initial exponential growth of an initial perturbation, was computed as a function of ΔT at fixed $P_0 = 173$ kPa for ΔT well in excess of the onset temperature. [Work supported by ONR.]

2:15

5PA4. Shape oscillations of bubbles driven by modulated ultrasonic radiation pressure: The complex frequency response. Thomas J. Asaki, Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814), and Eugene H. Trinh (Jet Propulsion Lab., Pasadena, CA 91109)

The shape of a bubble in water should change in response to the radiation pressure of an ultrasonic wave. Modulation of the radiation pressure at the resonance frequencies for shape oscillations should facilitate the stable excitation of such modes [P. L. Marston, *J. Acoust. Soc. Am.* **67**, 15–26 (1980)]. Levitation and detection of shape oscillations were previously discussed for bubbles in the size range 1–5 mm diameter [P. L. Marston *et al.*, *J. Acoust. Soc. Am.* **89**, 1885(A) (1991)]. Levitator design improvements have allowed stable trapping and shape oscillation detection for bubbles in the size range 1–12 mm diameter. The quadrupole mode was observed with resonance frequencies in the range 400–15 Hz. Oscillations over a more limited range were studied with a laser light scattering method that allowed the response phase and amplitude to be measured. The phase is referenced to the

amplitude modulation of the acoustic field. Further experiments have involved the effects of surfactants, levitation of thin liquid shells, splitting of bubbles, and surface instabilities. [Research supported by ONR and NASA.]

2:30

5PA5. Frequency of quadrupole mode oscillations of spheroidal liquid drops in an acoustic field. Xiaoyu Zheng, Yuren Tian, R. Glynn Holt, and Robert E. Apfel (Ctr. for Sonics and Ultrason., P. O. Box 2159, Yale Univ., New Haven, CT 06520)

Ultrasonic levitation of liquid drops in air has been used [Y. Tian *et al.*, *J. Colloid Interface Sci.* (submitted)] as a technique for measurement of the rheological properties (viscosity, interfacial tension, elasticity) of liquid surfaces and interfaces. The unavoidable deformation and asymmetry caused by the intensity of the sound field needed to levitate millimeter-sized water drops in air in a 1-g environment gives rise to a change in the frequency of quadrupole mode shape oscillations from the spherical value. This complicates measurements of rheological properties. Thus the frequency change due to the interaction with the sound field has been modeled. Using a unique conservation of energy approach, the shift due strictly to geometrical deformation of arbitrary origin is derived. Then realistic values of deformation and corresponding radiation force for typical cases are considered, modifying the previously obtained frequency shift values. Results are compared to experimental values obtained for millimeter-sized drops levitated

ultrasonically in air in a 1-g field. [Work supported by NASA through JPL Contract No. 958722.]

2:45

5PA6. Finite static deformation and translational motion of an acoustically levitated liquid drop in air. Yuren Tian, Glynn Holt, and Robert E. Apfel (Ctr. for Ultrason. and Sonics, Yale Univ., 2159 Yale St., New Haven, CT 06520)

Acoustical levitation has been applied in studies of droplet dynamics and interfacial properties. On the ground, when the density difference between droplet and host medium is large, the intensity of the levitation sound field must be strong enough to overcome gravity. As a result, the droplet may significantly deviate from spherical shape. A numerical method to determine the finite deformation and location of an acoustically levitated liquid drop in air has been developed. Compared with the previous studies [P. L. Marston, *J. Acoust. Soc. Am.* **67**, 15–26 (1980) and H. W. Jackson *et al.*, *J. Acoust. Soc. Am.* **84**, 1845–1862 (1988)], the interactions between droplet and sound field and the nonspherical acoustical scattering are included in our analysis, making the present method valid for droplets with aspect ratio as large as 2. With this method, the droplet shapes and locations as functions of sound pressure, surface tension, and droplet volume in both gravity and nongravity environments have been systematically calculated. The numerical results agree well with these experimental measurements and those of Trinh and Hsu [*J. Acoust. Soc. Am.* **79**, 1335–1338 (1986)]. [Work supported by NASA through JPL, Contract No. 958722.]

WEDNESDAY AFTERNOON, 13 MAY 1992

SALON F, 12:55 TO 4:30 P.M.

Session 5PP

Psychological and Physiological Acoustics: Peripheral and Central Physiology; Sensory Aids

Judith L. Lauter, Chair

Department of Communication Disorders, University of Oklahoma Health Sciences Center, 825 Northeast 14th, Oklahoma City, Oklahoma 73190

Chair's Introduction—12:55

Contributed Papers

1:00

5PP1. Inertial and geometrical properties of the human ossicles. R. M. Setser and R. D. Rabbitt (Dept. of Mech. Eng., Campus Box 1185, Washington Univ., St. Louis, MO 63130)

The principle volume moments of inertia and corresponding directions were numerically evaluated from three-dimensional reconstructions of the human ossicles. The location of the center of volume for each ossicle was also determined. Results were combined with average density measurements in order to estimate the principle mass moments of inertia. Surface data for the reconstructions were obtained in the form of parallel plane sections using computer-interfaced video microscopy.

Between 500 to 1500 (x, y, z) surface data points were recorded for each ossicle. Computation of the inertial and geometrical properties was greatly simplified through the use of integral kernels selected to reduce the volume integrals to surface integrals. Average results for two adult subjects provide the mass, volume, and principle mass moments of inertia for the malleus as $m = 25.0$ mg, $V = 13.5$ mm³, $I_1 = 94.1$ mg-mm², $I_2 = 9.1$ mg-mm², and $I_3 = 15.4$ mg-mm². For the incus $m = 25.7$ mg, $V = 15.5$ mm³, $I_1 = 119.8$ mg-mm², $I_2 = 107.8$ mg-mm², and $I_3 = 36.4$ mg-mm²; and for the stapes $m = 3.0$ mg, $V = 2.24$ mm³, $I_1 = 6.79$ mg-mm², $I_2 = 5.36$ mg-mm², and $I_3 = 1.82$ mg-mm². Ossicle reconstructions are graphically displayed to illustrate the directions of the principle axes and the location of the center of volume of each

ossicle. The computer-based surface models are also valuable in visualizing the geometry of individual ossicles and in providing geometrical data for use in middle ear models and prosthetic design. [Work supported by NSF BCS-8957206.]

1:15

SPP2. Cochlear fluid motion visualized with magnetic resonance imaging. Robert M. Keolian (Dept. of Phys. Code PH/Kn, Naval Postgraduate School, Monterey, CA 93943), Winfried Denk, Seiji Ogawa (AT&T Bell Labs.), and Lynn W. Jelinski (Cornell Univ.)

With a modification of standard magnetic resonance imaging (MRI) techniques, the oscillatory motion of water and cochlear fluids can be selectively imaged. Displacements of 160 nm could be detected (SNR = 3) with $33 \times 66 \times 300$ mm³ resolution in the cochlea of an anesthetized rat subjected to 4.6 kHz, 100-dB SPL sound, without averaging. The technique uses oscillating magnetic field gradients that are phase locked to the motion. The hydrogen nuclei of water that moves in synchrony with the oscillating gradient see on average a slightly different magnetic field than the static field, and accumulate an additional magnetic phase shift that can be imaged. This phase-sensitive-detection method, an NMR analog of the lock-in amplifier, permits three-dimensional detection of displacements far smaller than the spatial resolution. The method could be useful for speech research, in which vocalization could be used to drive the gradients, or in materials research, to map the spatial distribution of mechanical relaxations.

1:30

SPP3. Nonlinearities of mechanoelectric transduction in pigeon primary auditory fibers depend on the temporal pattern of their spontaneous activity. Andrei N. Temchin (Callier Ctr., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235)

Input-output functions (IOFs) were investigated in pigeons' auditory fibers with random and quasiperiodic spontaneous activity (SA) [A. N. Temchin, *J. Comp. Physiol. A* 163, 99-115 (1988)]. The first group showed regular sigmoidal IOF shape at the characteristic frequency (CF) and the slopes were virtually frequency-independent for each fiber. The second group had nearly linear IOFs only around CF and showed single-tone suppression of SA. At any suppressive frequency the IOF slopes were negative threshold and level at which total suppression occurred (15-20 dB). Positive slope of IOFs increased as the difference between CF and frequency of stimuli increased. Transition from total suppression of SA to saturation occasionally took place within as little as 10 dB. The minimal thresholds of suppression were often 7-8 dB less than thresholds at CF. The results cannot be explained by nonlinearities of basilar membrane and presumably indicate that active electrical properties of avian hair cells are involved in mechano-electric transduction. [Work supported by Callier-UTD Excellence in Education and M. F. Jonsson Funds.]

1:45

SPP4. Intensity coding using only low-spontaneous-rate auditory neurons. Fan-Geng Zeng (House Ear Inst., 2100 West Third St., Los Angeles, CA 90057)

A recent physiologically based model has shown that the optimal combination of rate information from both high-spontaneous rate (SR) and low-SR neurons predicts an intensity discrimination performance that exceeds human psychophysical performance across most of the normal range of intensity [I. M. Winter and A. R. Palmer, *J. Acoust. Soc. Am.* 90, 1958-1967 (1991)]. The present model, which uses only

low-SR neurons and a suboptimal decision process, predicts the human psychophysical data over a 100-dB range. Rate-intensity functions of six low-SR neurons are constructed based upon a linear relation between the neural threshold (dB) and the SR in logarithmic units. Variances of firing rates are derived from a Poisson process modified by a 2-ms dead-time. The decision variable is the unweighted sum of the firing counts from the six low-SR neurons. The same model has also been applied to the power law of loudness growth function, the midlevel hump on forward-masked intensity discrimination, and the effect of notched noise on intensity discrimination. [Work supported by NIH.]

2:00

SPP5. Influence of signal rise time on the amount of forward masking. Christopher W. Turner and Evan M. Relkin (Commun. Sci. and Disord., and Inst. for Sensory Res., Syracuse Univ., 805 S. Crouse Ave., Syracuse, NY 13244-2280)

Previously, a discrepancy between the amount of forward masking in the auditory nerve and published behavioral data was reported by these laboratories [E. M. Relkin and C. W. Turner, *J. Acoust. Soc. Am.* 84, 584-591 (1988)]; in that physiological experiment, however, a 2-ms signal rise time was employed, while the previous behavioral data used longer rise times. In the present behavioral study, both 2-ms and 10-ms signal rise-fall times were employed. The results show that for 10-ms ramps, the maximum amount of masking is large (at least 30 dB), in agreement with previous data. In contrast, for 2-ms ramps, the maximum amount of masking is smaller (15 dB). This is similar to the amount of masking in the auditory nerve reported in Relkin and Turner (1988), and indicates little or no discrepancy between behavioral and auditory nerve forward masking. This difference is observed for wide- and narrow-band maskers, suggesting that the results are not simply explained by the short-term spectra of the ramps. [Work supported by NINCDS.]

2:15

SPP6. Acoustic response properties of single units and unit clusters in the midbrain of the goldfish. Z. Lu and R. Fay (Parmly Hear. Inst. and Dept. of Psychol., Loyola Univ. of Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

Single units and small unit clusters were recorded from the torus semicircularis (TS) of the goldfish using indium electrodes. Response areas (RA) were formed as iso-level, spike count functions (75 to 1250 Hz) at 5-dB intervals throughout the unit's dynamic range. Spike count versus level functions and tuning curves were obtained from the RAs, giving characteristic frequency (CF), best sensitivity (BS), and Q_{10dB} . Peri-stimulus time (PSTH), interval, and period histograms were determined at CF and other frequencies. Units fall into three clusters with CFs at about 150, 400, and 1000 Hz, as is the case for saccular afferents. Tuning tends to be sharper in the TS than in the periphery. Lowest BS as a function of CF corresponds closely to the behavioral audiogram. Inhibitory effects are suggested by nonmonotonic level functions and by suppression of spontaneous activity. Slopes of spike count versus level functions are steep near CF and much shallower above and below CF. The first units encountered in a dorso-ventral electrode track tend to be tonic, short latency, and phase lock as accurately as saccular afferents. These are often two-unit clusters with different CFs but with phase-locked spikes 180° out-of-phase. Encountered next are more phasic, longer-latency responses with little phase locking, and occasionally with complexly shaped PSTHs. The frequency at which iso-level functions peak is generally independent of level, unlike saccular afferents. Thus frequency is far better represented by spike rate in the TS than in primary afferents. [Work supported by a Center Grant from NIDCD.]

2:45

5PP7. The effect of click rate on ABR stability measured with the REPs/ABR protocol. Judith L. Lauter, Lyman K. Hawkins, and Theodore H. Venema (Dept. of Commun. Disord., 825 N.E. 14th, Univ. of Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

Previous studies reported to this Society regarding our repeated evoked potentials (REPs) protocol for auditory brainstem responses (ABRs) have used a click rate of approximately 11 per s. Revisions in the REPs/ABR protocol requiring collection of as many as 16 waveforms in a single session encourage test streamlining, such as increasing click rate. The ABR literature suggests that while click rates of 30/s and above may affect 11/s-based absolute ABR parameters such as peak latency and peak amplitude, fewer changes will be seen with rates lower than 30/s. There are no data on the effect of click rate on ABR peak stability, the dependent variable targeted by the REPs procedure. Eight young adults with normal hearing (four women and four men) were tested in a within-subjects five-session design. Each session involved collection of 4 left ear, 4 right ear, and 8 binaural ABR waveforms; in sessions 1 and 5 a click rate of 11.1 per s was used; for sessions 2, 3, and 4, responses were evoked at click rates of 22.2, 33.3, and 44.4 per s, respectively. Results indicate that click rates faster than 30/s do create the most changes in 11/s-based peak latency and amplitude, though the effects for all increases are differentially distributed by peak, ear condition, and individual subject. Changes in click rate also have marked effects on ABR stability, both latency and amplitude, with details highly specific to individuals. In general, however, changes in stability as a function of click rate occur in the context of replication of the overall shape of the "stability profile" for each subject, providing further evidence of the "fingerprint" nature of this means of characterizing individual auditory brainstems.

3:00

5PP8. Efficacy of long latency auditory evoked/event-related potentials (AEP) in phonetic perception tasks. Lawrence F. Molt (Dept. of Commun. Sci. and Disord., Univ. of Georgia, Athens, GA 30602) and Richard S. Saul (East Tennessee State Univ., Johnson City, TN 37614-0002)

Latency functions for the auditory evoked potentials N100, P200, and the auditory event-related potential P300 were explored to determine their efficacy as indicators of neurologic processing time for the recognition of differences in three divergent levels of acoustic phonetic patterns. Stimuli consisted of digitally reproduced and altered pairs of spoken numbers with either acoustically identical (four, five with identical /f/), acoustically similar (four, five with natural /f/), or acoustically dissimilar initial phonemes (eight, two). Each stimuli pair was presented in a traditional 80%/20% AEP "oddball" paradigm. For the experimental group of 20 normal-hearing adult subjects, no statistically significant differences were observed in N100 or P200 waveform latency among the three levels of acoustic divergence. P300 latency exhibited significantly greater intersubject variability as phonetic acoustic divergence increased. Suggestions of possible brainstem or subcortical processing aspects, as well as subject-variable alternative processing strategies, are presented.

3:15

5PP9. The relationship between behavioral and auditory brainstem response thresholds in infants and adults. Lynne A. Werner, Lisa Rickard Mancl, and Richard C. Folsom (Dept. of Speech and Hear. Sci., Univ. of Washington, WJ-10, Box 47, Seattle, WA 98195)

It has been previously reported [L. A. Werner *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **88**, S18 (1990)] that for 3-month-old infants, there is a modest but significant correlation between detection thresholds measured behaviorally and using the auditory brainstem response (ABR) for tone pips at 4 and 8 kHz, but not at 1 kHz. It has been hypothesized that one should not see a correlation between behavioral and ABR thresholds among young normal-hearing adults, since variability in hearing levels in this group would be small, and that as infants' thresholds approached those of adults (around 6 months of age), the correlations we observed at 3 months would decrease. In the current study, detection thresholds were obtained from 20 to 30 yr olds at 1, 4, and 8 kHz and from 6-month olds at 8 kHz using behavioral and ABR measures to test these hypotheses. The stimuli were tone pips with 2-cycle rise and fall and no steady state. ABR thresholds were measured by reducing the level from 20 dBnHL until no response was observed, then increasing the level until the response reappeared. Behavioral thresholds were estimated adaptively, and defined as the average of the last 8 reversals obtained. For infants, a trial was considered "correct" if an observer said "yes" when a signal was presented to the infant, or "no" when no signal was presented. The observer had no prior knowledge of trial type. Adults were instructed to respond whenever they heard the tone pips. The correlation between ABR and behavioral threshold for adults was not significant at 1 or 4 kHz. Similarly, there was no correlation between the two measures for 6-month olds at 8 kHz. The correlation between ABR and behavioral thresholds for adults was significant, however, at 8 kHz. These findings suggest that variability associated with development of the auditory system is responsible for the ABR-behavioral threshold correlation seen at young ages. [Work supported by NIH DC00520.]

3:30

5PP10. Stimulus presentation formats in training paradigms for tactile aids. Janet M. Weisenberger (Div. of Speech and Hear. Sci., Ohio State Univ., Columbus, OH 43210)

Previous studies involving tactile aids for speech perception conducted in our laboratory have typically employed a training paradigm consisting of initial training with single-item speech stimuli in a citation-style live-voice presentation, followed by later training with connected discourse. This selection of training procedures was based on the assumption that for recognition of vibratory patterns corresponding to particular phonemes and/or syllables, such initial single-item presentation promoted rapid learning. However, given the sometimes substantial differences in the acoustic waveforms of items presented citation style or in the context of connected speech, it is possible that early initial training with citation-style stimuli would actually impair later performance with connected speech. Thus, although high levels of performance for users of tactile aids with connected speech have previously been obtained, it is possible that initial training with stimulus items in carrier phrases would be superior to citation-style presentation. In the present study, two groups of subjects were trained to use the Queen's University tactile vocoder; one group was trained initially with single-item stimuli presented citation style, and the other group was trained initially with stimuli presented in carrier phrases. Both groups were later tested in a

connected discourse tracking task. Preliminary analysis of the results suggests no differences between the two groups, suggesting that citation-style presentation and carrier-phrase presentation have equivalent effects on later connected speech performance. [Work supported by NIH.]

3:45

5PP11. Study of adaptive feedback stabilization of hearing aid. A. Maynard Engebretson and Marilyn French-St. George (Central Inst. for the Deaf, 818 S. Euclid Ave., St. Louis, MO 63110)

Instability is a problem with high-power and ITE hearing aids. Gain margins can be improved by 15 dB using an adaptive LMS filter to estimate the feedback characteristic and actively cancel it [Engebretson *et al.*, "Adaptive feedback stabilization of hearing aids," Second International Workshop on Hearing Impairment and Signal-Processing Hearing Aids, London, England, June 1991]. This algorithm was evaluated with a wearable digital hearing aid (WDHA). Nine hearing-impaired subjects were tested at 55 dB SPL, in quiet and in multitalker babble (6-dB SNR), with their own aid and with the WDHA programmed, with and without feedback stabilization (FBS), to match the REAR of their own aid. Subjects adjusted overall gain to their liking for each condition. Real ear measurements were taken to verify REAR. Correlation of word identification scores (PHFWL) with the subject's own aid and the WDHA (FBS disabled) was high. With FBS enabled, subjects chose additional gain that ranged from 2 to 10 dB and scores improved by 7 percentage points on the average across subjects. [Work supported by DVA.]

4:00

5PP12. Objective measures of hearing loss compensation techniques. Apichat Tungthangthum and Janet C. Rutledge (Elec. Eng./Comput. Sci. Dept., Northwestern Univ., Evanston, IL 60208)

Many techniques have been developed to compensate for hearing losses. With each new implementation, a subjective test was needed to verify its quality. In this study, objective measures to predict the results of a subjective test have been developed. A database of results from subjective testing based on four compensation techniques, multichannel

linear gain, multichannel compression, TVFD [J. C. Rutledge and M. A. Clements, Proc. IEEE ICASSP, 3641-3644 (1991)], and flat gain amplification, were used to develop the objective measures. The measures can be computed directly from the original speech waveform and a speech waveform processed using the compensation technique. Some parameters such as the energy in each band are measured from the original speech waveform and used as the input to a regression analysis model to calculate the coefficients of the model. Then the same type of parameters are measured from the processed speech waveform with some constraints of the hearing loss, such as hearing threshold, incorporated. The objective measures will be used to predict the performance quality of new techniques that are being developed. Subsequent subjective tests will then be used to fine tune the objective measures. It is hoped that the use of these objective measures will streamline the process of bringing new compensation techniques into fruition.

4:15

5PP13. The improvement of hearing ability after stimulating human cochlear region with low-intensity focused amplitude-modulated ultrasound. Shengke Zeng and Richard B. Beard (Biomed. Eng. and Sci. Inst., Drexel Univ., Philadelphia, PA 19104)

The experiment was designed to investigate the effect of low-intensity ultrasound on audibility after stimulating the human cochlea for a short radiation period [S. Zeng, M. S. thesis, Drexel Univ. (1990)]. The ultrasound carrier was focused, and amplitude modulated by an auditory signal. The intensity of the ultrasound in the cochlear region was no higher than 104 mW/cm² SPTP or 15.6 mW/cm² SPTA. The total radiation period was 30-60 min. Direct focusing of ultrasound on the cochlea was verified by observation of the subject's ultrasound auditory sensation. This sensation was stimulated by the auditory signal, demodulated from the ultrasound carrier in the cochlear region [I. A. Vartanyan and E. M. Tsirol'nikov, Fiz. Cheloveka 11 (3), 386-394 (1985)]. The audibility threshold of the subjects was tested before and after the modulated-ultrasound radiation. After the radiation, a decrease of the audibility threshold was observed. This audibility improvement was frequency dependent. For normal-hearing subjects, the improvement was 7.4 dB (mean) at 125 Hz and decreased linearly to 0.8 dB (mean) at 4 kHz. Above 4 kHz, the improvement was not significant. [This research was partially supported by Electro-Stim Corp.]

WEDNESDAY AFTERNOON, 13 MAY 1992

SALON H, 1:30 TO 3:00 P.M.

Session 5SA

Structural Acoustics and Vibration: Wide Band/Short Pulse Signal Interaction with Submerged Structures II

Leopold B. Felsen, Chair

Department of Ocean Engineering, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Contributed Papers

1:30

5SA1. Observation of the midfrequency enhancement of tone bursts backscattered by a thin spherical shell in water. Gregory Kaduchak

(Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A prominent feature predicted for the backscattering by thin spherical shells is an enhancement of the guided wave contributions in the

midfrequency range [L. G. Zhang *et al.*, *J. Acoust. Soc. Am.* **91**, 1862–1874 (1992)]. This feature may be useful for certain inverse problems and is associated with a strongly coupled slightly subsonic wave denoted by some authors as the a_0_- wave. The present research gives a comparison between a ray analysis and experiments in which tone bursts having carrier frequencies $\omega = kc$ in the range $35 < ka < 70$ were incident on an empty stainless steel shell in water. The sphere's radius to thickness ratio is $a/h = 43.8$. Time records of echoes provided a means for measuring the guided wave contribution relative to that of the specular reflection. As predicted the guided wave echo can be over three times larger than the specular echo and the ka dependence generally follows the predicted hump. [Work supported by ONR. The author acknowledges the advice of P. L. Marston.]

1:45

5SA2. Impulse response for backscattering by a thin spherical shell displayed in time and frequency. D. H. Hughes (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Time-frequency resolutions of the acoustical scattering from submerged targets utilizing exact form functions have been given by Yen *et al.* [*J. Acoust. Soc. Am.* **87**, 2359–2370 (1990)]. Spectrograms of back-scattered signals from a thin spherical are computed in the present research as smoothed Wigner distributions employing an algorithm given by Nuttall [NUSC Technical Report 8785 (1990)]. For the response to an impulse, the underlying dynamics are discussed in light of recent investigations by several authors working from different but complementary viewpoints. A deficiency in the spectrogram's ability to simultaneously resolve the instantaneous time-frequency content of back-scattered signals is exploited to display either the underlying resonant structure of steady-state scattering or the transient response to wide-band impulses or tone bursts. The spectrograms identify features associated with the midfrequency enhancement present in corresponding time-domain plots given by L. G. Zhang *et al.* [*J. Acoust. Soc. Am.* **91**, 1862–1874 (1992)]. Prompt features associated with the high-frequency enhancements are also evident. [Work supported by ONR.]

2:00

5SA3. Diffraction of evanescent wave tone bursts on membrane in air and related observations of bursts guided by plates in water. Thomas J. Matula (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

A previously described electromagnetic acoustic transducer (EMAT) was used to generate subsonic wave tone bursts on membranes [T. J. Matula and P. L. Marston, *J. Acoust. Soc. Am. Suppl.* **1** **88**, S167–S168 (1990)]. In the present research, the evanescent wave field in the air surrounding the membrane was studied. Diffraction of the burst by a sharp edge in the air was also observed. The diffracted signal was measured using a microphone as a function of the gap h between the membrane and a razor edge. The diffracted pressure decreases exponentially with increasing h as expected from an approximate analysis of edge diffraction of evanescent waves [L. B. Felsen, *J. Opt. Soc. Am.* **66**, 751–760 (1976)]. The importance of such diffraction is that it produces far-field radiation of energy that otherwise would be trapped to a membrane or plate in the absence of scattering. In related work an EMAT is used to generate tone bursts of bending waves on an aluminum plate. The bursts propagate down the plate into water where the surrounding wave field is probed. [Work supported by ONR. The author acknowledges the advice of P. L. Marston.]

2:15

5SA4. Comparison of acoustical images with cylindrical holograms. Charles F. Gaumond, Angie Sarkissian, Earl G. Williams (Naval Res.

Lab., Washington, DC 20375-5000), and Timothy J. Yoder (SFA, Inc., Landover, MD 20785)

For this comparison, space-time images are generated from broadband bistatic scattering data. Two imaging methods are used: an extension of reflection tomography [P. B. Abraham and C. F. Gaumond, *J. Acoust. Soc. Am.* **82**, 1303–1314 (1987)] and far-field cylindrical holography. The sharpness of the space-time impulse response, or resolution, as well as the appearance of scattering mechanisms such as reflection, diffraction, and plate wave propagation are shown for the case of a right circular cylindrical shell with hemispherical endcaps.

2:30

5SA5. Experimental evaluation of backscattering from finite internally loaded cylindrical shells. Charles N. Corrado, Jr., Matthew Conti, and Ira Dyer (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

Dynamic interaction between a shell and internal structural elements may be significant in backscattering. These interactions are of particular interest to us at frequencies where wavelengths are comparable to characteristic discontinuity lengths of the structure. Scattering by three cylindrical shell models has been measured to delineate regimes where the influence of internal structures is significant, and to highlight fundamental backscattering processes common to more complex structures. The models are: an empty shell, a duplicate empty shell with unequally spaced ribs, and a duplicate ribbed shell with resiliently mounted, wave-bearing internal structural elements. The measurements were conducted with wide-band pulses covering the frequency band of $2.75 < ka < 10.5$ corresponding to 3/4 to 3 times the ring frequency of the empty shell. The internal structures are found to cause significant changes in target strength at all angles of incidence away from beam aspect, and throughout the frequency range studied. The backscattered signature in the time domain shows an increased initial peak response induced by the internal structures, as well as group delay and decay rate properties induced by shell discontinuities and by interaction between the shell and the internal structures. [The authors acknowledge valuable inputs from J. R. Fricke and Y. P. Guo of MIT, and NRL for acquisition of the backscatter data. Research sponsored by ONR.]

2:45

5SA6. Windowed Fourier transforms—Application to structural acoustics. John J. McCoy (School of Eng. and Architecture, Catholic Univ. of America, Washington, DC 20064)

The windowed Fourier transform provides the resolution of a (space/time) forcing of a linear dynamical system, using a set of base functions spanning a (space-wavelength/time-frequency) phase space. The elements of the set of base functions, separately chosen as fundamental forcings of the dynamical system, define a set of fundamental solutions from which the system response to an arbitrary forcing can be obtained by a process of synthesis. A specific characterization of the nature of the windowing, i.e., of the set of base functions for resolving the forcing, leads to a characterization of the set of fundamental solu-

tions resulting therefrom, as a specific "species" of response functions. The recent experience of a number of different researchers of a broad range of problems is that one can achieve both physically intuitive and computationally efficient solution algorithms by choosing phase-space base functions that remain localized under the operation that obtains a system response from a system forcing. Three examples of application of these ideas will be presented. One is for the range evolution of a narrow-

band acoustic field emanating from a source plane by a process of (physical space/direction) phase-space marching. The second is the propagation of a wideband signal through a dispersive waveguide. The third is the acoustic field radiated from a submerged plate subject to a forcing of compact support in both space and time. [Work supported by ONR.]

WEDNESDAY AFTERNOON, 13 MAY 1992

SALON E, 1:00 TO 3:05 P.M.

Session 5SP

Speech Communication: Cross-Language Studies

James E. Flege, Chair

Department of Biocommunication, University of Alabama, Birmingham, Alabama 35294

Chair's Introduction—1:00

Contributed Papers

1:05

5SP1. Coarticulation in Russian unstressed vowels. David K. Evans-Romaine (Prog. in Linguist., Univ. of Michigan, 1076 Frieze, Ann Arbor, MI 48109)

Studies of vowel-to-vowel coarticulation in Russian have generally found vowels in VCV sequences to have little influence on each other's articulation. This has been attributed to Russian's contrastive palatalization, which is assumed to block coarticulation by preventing the pharynx and tongue body from moving smoothly from one vowel configuration to the next during stop closure. If this is correct, coarticulation should be minimal even for reduced vowels. This hypothesis was tested by examining coarticulation in the nonfront reduced vowels of Russian. Here, $F1$ and $F2$ were measured at the onset, center, and offset of five repetitions of prestress /o/, poststress /o/, and stressed /a/ produced by native speakers in /tVb__b^(j)Vt/ and /pVd__d^(j)Vt/ nonsense words. Posttarget consonants were plain or palatalized; flanking vowels were /i,a,u/, symmetrical. All speakers showed substantial coarticulation throughout the poststress target, during the first half of prestress target, and in the onset of stressed target. These results suggest that contrastive palatalization does not preclude significant vowel-to-vowel coarticulation.

1:20

5SP2. Segmental phonetic cues to language identity. James Emil Flege and Murray J. Munro (Dept. of Biocommun., Box 503 UAB Sta., Birmingham, AL 35294-0019)

In experiment 1, detailed acoustic analyses of 42 tokens of *taco* as spoken either in Spanish or English by monolinguals, and in both languages by early and late learners of English L2, were performed. As expected, the early learners' renditions of *taco* in Spanish and English differed more than the later learners'. Correlation techniques revealed that the extent to which the bilingual talkers approximated English phonetic norms for the four segments in *taco* was interrelated, suggesting that the word is an important unit in cross-language phonetic interference. In experiment 2, three expert listeners identified the language in

which the *taco* tokens had been spoken, then rated each for goodness. The Spanish and English monolinguals' renditions of *taco* were readily differentiated, indicating that purely phonetic differences can cue language identity. The late learners' English and Spanish renditions were judged to be only slightly different whereas the listeners tended to judge the early learners' renditions as being clearly Englishlike or Spanishlike. Multiple regression analyses examining the acoustic variables from experiment 1 were able to account for most variance in the listeners' identification judgments and goodness ratings (81%, 97%). In experiment 3, three acoustic parameters identified as significant predictors of the listeners' perceptual judgments (viz., VOT of "t," the spectral quality of "a," and "o" duration) were varied from "Spanish" to "neutral" to "English" in synthetic versions of *taco*. All other parameters were set to the average of values observed for the Spanish and English monolinguals. The hypothesis was tested that language identification may be based on a single segment, whereas goodness ratings are based on properties distributed through the entire word.

1:35

5SP3. Phonetic underspecification in Marshallese. John D. Choi (Dept. of Linguistics, Univ. of California—Los Angeles, Los Angeles, CA 90024-1543)

The vowels of Marshallese have been analyzed as contrasting only along the height dimension, variation along the front-back dimension being predictable by the secondary articulations of adjacent consonants. Spectrographic evidence suggests this analysis is correct and that the surface allophones can be derived by interpolating between C_1 and C_2 targets across a vowel that is unspecified for $F2$. To test this hypothesis, /C₁VC₂/ words ($C_n = \{/p^h/, /p^y/, /t^h/, /t^y/\}$ and $V = \{/i/, /e/, /a/\}$) were recorded from four speakers. Then $F2$ was measured from vowel onset to offset. Preliminary findings based on two of the speakers shows that while $F2$ variation at the vowel center is statistically significant as a function of vowel category, the proportion of variance accounted for by vowel category is very low compared to that accounted for by the secondary articulation of C_1 and C_2 . This is interpreted as evidence that a vocalic $F2$ target may not be necessary in modeling the trajectory.

Multiple linear regression tests support this interpretation, showing that the vowel midpoint can be accurately predicted without reference to a vowel target. Regression analysis was performed on 15 time-normalized points with and without an intercept k , where k represents an independent vowel contribution to F_2 . The results show that the contribution of k is negligible (± 0.001). Data from the two remaining speakers will be incorporated and discussed in terms of phonetic underspecification in a target-interpolation model.

1:50

5SP4. Towa tones and stress. Alan Bell (Dept. of Linguistics, Box 295, Univ. of Colorado, Boulder, CO 80309)

This paper reports on the phonetics of tones and stress in Towa, a Kiowa-Tanoan language of New Mexico. Towa words have an initial prominent syllable bearing either a high tone or a high falling tone. Long and short vowels also contrast in these initial syllables. Succeeding syllables, which permit only short vowels, may have high (after initial high only), mid, or low tone. (Words appear to have only two underlying tonal melodies, H or HL , with the L falling either on the first or the second syllable.) The basic realizations of initial high and falling tones resemble the Mandarin Chinese first and fourth tones. The contrast is reduced in short monosyllables, and there are notable contextual assimilatory effects. The main issue is whether initial syllable prominence should be attributed to the initial high pitch of words or to stress. On the basis of comparative durations, vowel quality reduction, and the phonetic consequences of prominence loss in compounds, incorporations, and successive initial syllables, it is argued that Towa initial syllables are stressed.

2:05

5SP5. Segment and syllable frequency. Ian Maddieson (Dept. of Linguistics, UCLA, Los Angeles, CA 90024-2543 and Univ. of California—Berkeley, Berkeley, CA 94720)

The frequency of individual segments and of each particular syllable has been counted in lists of about 2000 to 5000 lexical entries in a sample of languages drawn from the major areal/genetic groupings. An analysis of a subset of these languages with small segment inventories and simple phonotactics (Maddieson and Precoda, 1992) has shown that the frequency of a particular syllable type can be better predicted from the overall segment frequencies than from the assumption that adjoining articulatory similar segments are preferred, as suggested by Janson (1986). Additional languages are being added to the database to test the generality of this conclusion over a more diverse set of languages. The extended set includes Chinese, Polish, Korean, Comanche, Kwakw'ala, Yupik, !Xóǝ, Ngizim, Maninka, Gbaya, Fasu, Turkish, as well as Hawaiian, Pirahã, Rotokas, Shipibo, and Kadazan which were reported on earlier.

2:20

5SP6. Effects of implosives on F_0 in SiSwati. Richard Wright and Aaron Shryock (Phonetics Lab., Dept. of Linguistics, UCLA, 405 Hilgard Ave., Los Angeles, CA 90024-1543)

As is widely known, the fundamental frequency (F_0) of the onset of a vowel after voiceless consonants is significantly higher than after voiced consonants. It has been assumed in the literature that implosives (glottalic ingresses) have a raising effect similar to voiceless stops. To

investigate this hypothesis, a study was conducted examining the effects of implosives in SiSwati. Four subjects, three males and one female, participated in the study. The wordlist consisted of eight CVX tokens, where $C = /p^h, \delta, m, m/$ (voiceless aspirated, implosive, breathy, plain voiced) and where $V = /a/$ with high or low tone. The tokens were spoken in the frame: "Tsáni ___ fùtsi." (Say ___ again). The results for $/p^h/$ duplicate the results of earlier studies showing a higher onset F_0 . The nasals, $/m, m/$, lowered onset F_0 . However, the results for $/\delta/$ indicate that implosives have no significant raising or lowering effect on F_0 .

2:35

5SP7. Focus in Peninsular Spanish. Guillermo Toledo (Univ. de Barcelona, Barcelona, Spain, and Lab. de Investigaciones Sensoriales, CONICET, CC53, 1453, Buenos Aires, Argentina) and Eugenio Martínez Celdrán (Lab. de Fonètica, Divisió I, Facultat de Filologia, Univ. de Barcelona, Gran Via de les Corts Catalanes, 585, 08007, Barcelona, Spain)

This study explored the acoustic realization of focus in several dialects of Mediterranean Peninsular Spanish. The materials consisted of two corpora of declarative sentences with differences of neutral versus broad versus narrow focus and neutral versus single versus dual focus triggered by wh-question contexts. For that goal values of focal prominence in three prosodic parameters, F_0 , duration, and amplitude, were investigated through intonation contours and digital spectrograms of seven male speakers' emissions. The statistical analysis of the measurements has shown no acoustic differences among focused and unfocused items, and among sentences in the two corpora. The results provided support for prior studies in American and Canary Spanish which were undertaken through similar experimental designs. However, these data appeared to contradict the general findings reported elsewhere which have shown a higher degree of prominence in the marking of focus both in cognate languages like French and Italian and in Germanic languages.

2:50

5SP8. Focus in insular Spanish. Josefa Dorta (Lab. de Fonètica, Univ. de La Laguna, Campus Central, La Laguna, 38291, Tenerife, Canary Islands, Spain) and Guillermo Toledo (Univ. de La Laguna, Tenerife, Spain, and Lab. de Investigaciones Sensoriales, CONICET, CC53, 1453, Buenos Aires, Argentina)

An acoustical study of linguistic focus in two experiments of production through materials emitted by seven Canary Islands speakers was explored. In one experiment different wh-question contexts triggered read emissions of neutral, broad-focus, and narrow-focus sentences. In a second experiment, speakers produced a read corpus of neutral, single focus at first and last position in the utterance, and dual-focus sentences contextually designed. The experiments have been set up to analyze prominence in focused items encoded through F_0 , duration, and amplitude features. To that end, intonation contours and digital spectrograms were made for acoustic measurements. Results did not support the findings reported for other cognate languages as Italian and French which indicated a higher degree of prominence in focused words. On the contrary, the statistical data showed no differences among focused and unfocused items, and among neutral and focused sentences. The findings suggested that in this modality of Spanish the prosodic encoding of focus appears to be ruled by the speakers in a contextually independent and unpredictable manner.

Session 5UW

Underwater Acoustics: Propagation II

John S. Perkins, Chair

Naval Research Laboratory, Code 5160, Washington DC 20375-5000

Chair's Introduction—12:55

Contributed Papers

1:00

5UW1. Travel time corrections for ice scattering in the 1988–1989 Greenland Sea tomography experiment. J. Lynch, R. Pawlowicz, N. Beliakova (Woods Hole Oceanog. Inst., Appl. Ocean Phys. and Eng. Dept., Woods Hole, MA 02543), P. Worcester (Scripps Inst. of Oceanog.), and Jin Guoliang (Shanghai Acoust. Lab.)

Previous work by Guoliang and Wadhams [Prog. Oceanog. 22, 249–275 (1989)] has shown that reflections of acoustic rays by a sea ice cover can significantly affect acoustic travel times in an ocean tomography experiment. This can lead to warm or cold biases in the ocean inversions, if unaccounted for. This paper presents a discussion of how one deals with correcting the travel times of acoustic arrivals seen in the 1988–1989 Greenland Sea tomography experiment for ice effects. The discussions will include: generalization of the beam displacement formalism of Guoliang and Wadhams, determination of the reflection coefficient versus incidence angle of the rays using scattering amplitude data, estimation of beam displacements and time delays, and effects of the corrections on inversions for the ocean temperature field. Directions for future work will also be discussed. [Work supported by ONR and NSF.]

1:15

5UW2. Approximate methods for reflection from solid bottoms. Z. Y. Zhang and C. T. Tindle (Phys. Dept., Univ. of Auckland, Auckland, New Zealand)

The acoustic reflection coefficient for a solid ocean bottom can be well approximated by assuming an "equivalent fluid" bottom of suitably chosen parameters. The density and attenuation of the equivalent fluid are different from the corresponding parameters of the solid. In particular, the density is substantially smaller and it is shown that simple neglect of the shear wave parameters of the solid can give poor results. The equivalent fluid can be used for both normal mode and ray calculations of the field in the water and gives essentially the same results as calculations that include the full shear waves effects. For isovelocity water, bottom reflection can also be modeled by assuming a perfect reflector at an "effective depth." A new complex effective depth is defined to account for energy loss due to attenuation and the generation of shear waves. The resulting normal mode parameters are also complex. The sound field in shallow isovelocity water can be found very accurately and rapidly using the complex effective depth approach.

1:30

5UW3. The effective depth approximation for sound propagation in shallow water over a sediment layer and a hard rock basement. Stewart A. L. Glegg (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

This paper considers an effective depth approximation for an ocean waveguide with a bottom in which a sediment layer overlies a hard rock basement. The primary application of this approximation is to the calculation of three-dimensional propagation over sloping bottoms. It is shown that at low frequencies the resonances in the sediment layer cause the effective depth to oscillate rapidly. At high frequencies, when the attenuation in the sediment layer becomes significant, the effective depth tends to zero and the acoustic propagation is dependent on the modal attenuation. This can be incorporated into the effective depth theory by allowing the effective depth to be a complex number, defined in terms of the impedance of the water/sediment interface. Comparison with exact calculations shows that the effective depth approach gives a good approximation for the lower-order modes, but does not include the effects of the higher-order modes which are shown to make a significant contribution to the fine scale structure of the long range transmission loss. [Work supported by ONR.]

1:45

5UW4. Harmonic responses in a multilayered saturation porous media. M. K. Kuo, C. S. Yeh, and J. W. Liaw (Inst. of Appl. Mech., Natl. Taiwan Univ., Taipei, Taiwan 106, Republic of China)

The time harmonic Green's function for a point force in a multilayered fluid-saturated porous media is derived in the context of Biot's theory. The crucial step of the analysis is a global matrix technique, which is different from the propagator matrix approach [T. Yamamoto, J. Acoust. Soc. Am. 73, 1587–1596, 1599–1620 (1983)] and the generalized ray expansion via Gauss–Seidel matrix iterative method [R. C. Y. Chin *et al.*, Wave Motion 7, 43–65 (1985)]. The essence of the method is the use of the normal and tangential components of solid displacement and the normal component of fluid displacement at each interface (including the outer surface) as basic unknowns in the wave-number domain. The needed continuities on displacement components across interfaces of the multilayered medium is then ensured automatically. These as yet undetermined interface displacements are finally determined from the boundary conditions at the outer surface of multilayered medium and the continuities of tractions across interfaces. The resulting linear system of equations is straightforward assembled and manifested in a global form. Finally, the wave-number integrals are evaluated numerically to yield the frequency responses. The numerical results are presented for the cases of a point source in a multilayered ocean with multilayered fluid-saturated porous seabeds.

2:00

SUW5. The influence of seabed properties on the generation and propagation of Scholte interface waves. Hassan B. Ali (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Scholte seismic interface waves can be an important component of very low-frequency (VLF) acoustic propagation, particularly in shallow-water environments. Under certain conditions, these waves may provide the only effective mechanism for the propagation of VLF energy. Moreover, they are often a significant, if not dominant, contributor to the VLF/ULF ambient noise field, both seismic and waterborne components. This work examines the generation and propagation mechanisms of Scholte waves using the results of recent NOARL measurements and new numerical predictions. It is shown that the propagation characteristics of these seismic waves, including their presence in the water column, are dependent not only on the bottom geoacoustics (particularly shear speeds) but also on the types and thicknesses of layers overlying the basement level. The measurement of Scholte waves even in the apparent absence of near-bottom sound sources suggests the excitation of secondary Scholte waves by the roughness of the seabed in the vicinity of the receiver. In particular, these secondary Scholte waves are attributed to the conversion into shear (*S*) waves of incident waterborne compressional (*P*) waves. Conditions favoring such *P* to *S* conversions, and the implications for acoustic propagation, are discussed. [Work supported by ONR.]

2:15

SUW6. Sensitivity of predicted shallow-water propagation loss to empirical and extrapolated bottom loss values. Joseph M. Monti, Jefferey H. Sammis, Peter D. Herstein, and David G. Browning (New London Detachment, NUWC-Newport Div., New London, CT 06320)

The majority of measured bottom loss values were obtained in deep water at grazing angles of 10 deg or greater. An empirical fit to these data, if extended smoothly down to small angles, results in a value

greater than 2 dB per bounce at 0 deg. Geophysical models suggest that for hard bottoms a critical angle would be reached in this low-grazing angle region and that the bottom loss would then drop sharply, reaching a zero value at 0 deg. Under downward refracting conditions in shallow water, low-grazing angle paths may provide, in many cases, the dominant propagation mode. Hence, the value of low-grazing angle bottom loss is critical especially under strongly downward refracting conditions. Following the example of Urick, an empirical bottom loss formula developed by Bell has been modified. At low-grazing angles, its regular value at 10 deg down to reach a zero value at 0 deg has been linearly extrapolated. An analysis is conducted for each formula (empirical and extrapolated) for several shallow-water locations and source-receiver configurations.

2:30

SUW7. Wave conversion in cross sections of a weakly inhomogeneous waveguide. Alexander S. Starkov (Dept. of Higher Math., Inst. of Refrigeration Industry, Lomonosova, 9, Saint Petersburg, 191002, Russia)

Acoustical waves propagation in a low-speed layer of variable depth based on a fluid bottom is considered. The depth and sound velocities in the layer and bottom vary slowly with horizontal distance. Traditional adiabatic expansion breaks down in the cutoff region. Here, the goal is to investigate wave conversion in tapered and broadened waveguides. In the first case, the lateral wave is created in every critical cross section [A. S. Starkov, *Akust. Zh.* 36, 348-354 (1990)]. Therefore, at cutoff the normal wave mode creates a secondary lateral wave. Similarly, in the case of a broadened waveguide the lateral wave creates a normal mode. The fields in the transition region are obtained with various asymptotic methods and described in terms of Airy and incomplete Airy functions. The coefficients of different waves excitation are calculated. It is shown that the curvature of the boundary leads to the more complicated Fock type expansions. The resulting wave fields are shown to have additional oscillations before cutoff and decrease slowly after cutoff.

WEDNESDAY AFTERNOON, 13 MAY 1992

SYMPHONY HALL, 5:00 TO 6:00 P.M.

Plenary Session

Eric E. Ungar, Chair

President, Acoustical Society of America

Presentation of Awards

R. Bruce Lindsay Award to Anthony A. Atchley
Gold Medal to Ira J. Hirsh

Session 6BV

Bioresponse to Vibration and Physical Acoustics: Acoustic Scattering in Biological Tissues

K. Kirk Shung, Chair

*Bioengineering Program, Pennsylvania State University, University Park, Pennsylvania 16802**Invited Papers*

8:15

6BV1. An overview of theoretical and experimental developments on ultrasonic scattering in tissues. K. Kirk Shung (Bioeng. Prog., 231 Hallowell Bldg., Penn State Univ., University Park, PA 16802)

The growth of ultrasonic imaging in the past 2 decades has been phenomenal because of its many advantages. It is noninvasive, capable of providing both anatomical and blood flow information in real-time, less expensive, and portable. It is plausible to see why ultrasonic imaging is now second only to x ray in terms of number of clinical procedures performed. Although it is widely used and much diagnostic information can be retrieved from ultrasonic images, the genesis of ultrasonic texture or speckle exhibited by a tissue in an ultrasonic image is still poorly understood. Since the ultrasonic image is formed from the echoes backscattered from tissues, a systematic and thorough study of ultrasonic scattering in biological tissues is crucial for a better interpretation of ultrasonic images and for the further development of this modality. It is for this reason that ultrasonic scattering in tissues has been under intensive investigation both theoretically and experimentally for many years. In this paper, important experimental approaches that have been developed to measure ultrasonic scattering from tissues and theoretical models developed to analyze these results will be reviewed. [Work supported by NIH Grant No. HL28452.]

8:45

6BV2. Clinical relevance of scattering. Gary A. Thieme (Dept. of Radiol., Milton S. Hershey Medical Ctr., Penn State Univ., Hershey, PA 17033)

Diffuse scattering is the basic sound interaction responsible for the soft tissue detail seen in modern clinical gray scale ultrasound imaging. Its counterpart is specular reflection which is the physical interaction responsible for the bright interfaces seen in bistable and modern images. This paper will illustrate the effects of operator-controlled system variables on the image generated from diffuse scattering processes and will demonstrate the clinical aspects of diffuse scattering through the use of tissue-mimicking phantoms and normal and pathologic clinical imaging examples. Four parameters (spatial resolution, contrast resolution, texture, and speckle) influence our ability to detect focal lesions and diffuse pathologic abnormalities in the clinical setting. Signal processing in the ultrasound system and adjustment of controls by the operator affect the perception of diffuse scattering. Imaging is influenced by transducer frequency and focusing characteristics, system gain, time gain compensation, system dynamic range, preprocessing functions, post-processing functions, system power, and frame averaging. Images illustrating the effects of these parameters will be shown.

9:15

6BV3. Acoustic scattering theories applied to biological tissues. David G. Brown (Ctr. for Devices and Radiol. Health, FDA (HFZ-140), 12720 Twinbrook Pky., Rockville, MD 20857) and Michael F. Insana (Univ. of Kansas Med. Ctr.)

Much information of potential diagnostic significance concerning acoustic tissue characteristics can be made available only through sophisticated signal processing. This requires a fundamental understanding of the basic acoustic interactions in tissue. The basic equations of acoustic scattering theory are reviewed as they relate to scattering in soft biological tissues. Using the inhomogeneous continuum model for scattering processes, it is shown how the basic scattering properties are derived for both coherent and incoherent scattering. The treatment begins with scattering from a single inhomogeneity and is then developed to cover scattering from random continua. Backscatter coefficients for isotropic random media are addressed and the extension to anisotropic media considered. An example is given of the analysis of backscatter measurements from the kidney.

6BV4. Measurement system effects in ultrasonic experiments. Robert C. Waag (Depts. of Electrical Eng. and Radiol., Univ. of Rochester, Rochester, NY 14627) and Jeffrey P. Astheimer (66 Sibley Rd., Honeoye Falls, NY 14472)

Arbitrary fields are represented as angular spectra of plane waves and arbitrary time waveforms as frequency spectra of temporal harmonics in an integral expression that includes instrument effects on the spectrum of the scattered pressure. A transformation of variables yields a form convenient for interpretation. Simplified expressions are developed for the effect of beam patterns when the detector time gate has an infinite duration. Simplified expressions are also developed for the effect of the emitter waveform and detector time gate using a narrow-band approximation and for the effect of the emitter beam and detector sensitivity patterns using a quasi-plane-wave approximation. The simplifications yield insightful forms for measurement system characteristics and these forms are evaluated analytically for Gaussian spatial apertures and Gaussian temporal waveforms. Numerical evaluations of general expressions for the weight of Gaussian, exponential, and uniform spatial apertures are also given. The results indicate instrumentation parameters that can be employed to measure intrinsic characteristics of a scattering medium and may be used to design experiments from which intrinsic parameters of scattering media are obtained.

10:15

6BV5. Characterization of cardiovascular tissue with ultrasonic backscatter. Samuel A. Wickline, Julio E. Perez, Benico Barzilai, Burton E. Sobel, and James G. Miller (Washington Univ. School of Medicine, Box 8086, 660 S. Euclid, St. Louis, MO 63110)

Quantitative characterization of cardiovascular tissue with backscattered ultrasound has evolved from a laboratory tool to a clinical adjunct for conventional echocardiographic imaging. Integrated backscatter imaging was developed by our laboratory to permit analysis of the physical properties of myocardial tissue independent of wall motion or chamber dimension. Initial observations in experimental animals and later in patients revealed that normal myocardium exhibits a cardiac cycle-dependent alteration in the intensity of backscatter that reflects intrinsic myocardial contractile performance. Detection of blunted cyclic variation of backscatter has been used by this laboratory and others to delineate regional contractile dysfunction in patients with infarction, "stunned" myocardium, cardiomyopathy (idiopathic and diabetic), hypertension, and cardiac transplant rejection. Automated ventricular boundary detection is another recent clinical application based on integrated backscatter imaging that allows on-line display and tracking of cavity edges throughout the cardiac cycle in either transthoracic or transesophageal imaging modes to define global ventricular performance. Another recent clinical innovation, "lateral gain compensation," permits quantification of and compensation for the ultrasonic anisotropy of scattering exhibited by myocardial tissue, which is attributable to a highly ordered and complex arrangement of myofibers. Promising novel developments for tissue characterization include determination of the frequency dependence of scattering, which may improve the specificity for echocardiographic diagnosis of acute infarction and cardiomyopathy and facilitate quantification of pathologic tissue architectural remodeling. [Work supported by NIH Grant Nos. HL42950, HL40302, HL17646.]

Contributed Papers

10:45

6BV6. Feature extraction from scattered acoustic fields. Kavitha Chandra (Dept. of Electrical Eng., Lab. for Advanced Comput., Univ. of Massachusetts—Lowell, Lowell, MA 01854)

The scattered pressure field from two-dimensional inhomogeneous biological media is first determined by direct solutions of the integral equation. This work deals with the analysis and interpretation of the scattered pressure measured at the observation locations. Deterministic as well as stochastic isotropic and anisotropic spatial distributions of scatterers are considered. Specific similarity groupings of the variables that allow a global medium characterization from the scattered pressure field are determined. Techniques for estimating the porosity of the scattering volume, the characteristic dimension of the scatterers, and the principal orientation of the distribution are presented. Special attention is paid to cases where the scattering volume is self-similar in the distribution of the radii of the scatterers. The influence of the scattering process on the magnitude of the characteristic fractal dimension of such a distribution is examined.

11:00

6BV7. Reflections and emissions from stones during electrohydraulic lithotripsy. P. R. Delmenico and R. D. Rabbitt (Dept. of Mech. Eng., Campus Box 1185, Washington Univ., St. Louis, MO 63130)

Lithotripsy is a noninvasive procedure to destroy biliary and renal concretions using shock waves generated externally to the patient and focused at the stone site. The effectiveness and safety of lithotripsy procedures are partially determined by the accuracy of stone targeting. The presentation will cover analysis of pressure waves reflected from the stone to determine stone location, degree of pressure wave focusing, and stone geometry (with changes in stone geometry providing a measure of stone destruction). In the experiments, a Wolf model 2137.50 electrohydraulic lithotripter is used in conjunction with an ellipsoidal focusing system. Piezoelectric pressure transducers within the focusing system provide a means to measure pressure waves both before and after reflection from a variety of targets. In addition, a computer simulation for axisymmetric shockwave propagation in an ellipsoid is developed to predict reflections from stones of arbitrary geometry and position relative to the far focus of the ellipsoid. This ray method solution of the high-frequency approximation to the wave equation accounts for the

formation of caustics, while other nonlinear effects are neglected. The simulation allows the visualization of time varying pressure fields within the lithotripter focusing system and around the targeted stone. This research could provide a starting point for a method of dose estimation in lithotripsy procedures. [Work supported by NSF, BCS-8957206.]

11:15

6BV8. Sizing of microbubbles by nonlinear scattering. Xucai Chen and Richard S. Meltzer (Dept. of Med. and Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Box 679, Rochester, NY 14642)

A technique to determine the size distribution of microbubbles in echo contrast agents is presented. The experimental apparatus consists

of three confocally positioned acoustic transducers, two of which serve as transmitters and the other serves as a receiver. One transmitter transmits tone bursts at a fixed frequency higher than the resonance frequencies of the microbubbles, while the other transmitter transmits tone bursts whose center frequencies are swept through the range of resonance frequencies of the microbubbles. Scattered tone bursts are detected by the receiver and recorded by a digital oscilloscope and a computer. A two-dimensional power spectrum is obtained by calculating the spectrum of the received signal as a function of the sweeping frequency. Amplitudes of the second harmonic of the sweeping frequency [D. L. Miller, *Ultrasonics* **19**, 217-224 (1981)] as well as the combination frequencies [V. L. Newhouse and P. Mohana Shankar, *J. Acoust. Soc. Am.* **75**, 1473-1477 (1984)] are used to calculate the size distribution of the microbubbles.

THURSDAY MORNING, 14 MAY 1992

SALON J, 8:30 A.M. TO 12:20 P.M.

Session 6EA

Engineering Acoustics: Otoacoustic Emission Techniques, Transducers, and Applications

Mead C. Killion, Chair

Etymotic Research, Elk Grove Village, Illinois 60007

Chair's Introduction—8:30

Invited Papers

8:35

6EA1. The origin of acoustic distortion in the cochlea. A. M. Brown and S. A. Gaskill (Lab. of Exp. Psychol., Univ. of Sussex, Brighton BN1 9QG, England)

Two-tone stimulation of the ear is a powerful tool for investigating cochlear function and is gaining increasing acceptance as an audiological test. The intermodulation distortion generated by two tones includes numerous cubic components. The behavior of cubic lower sidebands has been studied by measuring them with a microphone in human and rodent ear canals as the stimulus frequency was changed at various stimulus levels. It was found that distortion produced by low level stimuli (level of f_2 at or below 40 dB SPL) is due to activity at the f_2 region on the cochlear partition, and that the sound measured in the ear canal has passed through an additional cochlear filter (possibly the tectorial membrane). The picture may be complicated in the human ear by added energy from the distortion product frequency region. At high stimulus levels, the response is more difficult to interpret. There may be distinct sources of $2f_1 - f_2$ caused by secondary modulation of f_1 by $f_2 - f_1$ ($f_2 - f_1$ belongs in the quadratic, rather than the cubic family). This interaction would also give f_2 (thus enhancing or suppressing the f_2 level, depending on phase relationship) and upper sidebands, such as $2f_2 - f_1$.

9:15

6EA2. Otoacoustic emissions and auditory perception. Glenis R. Long (Dept. of Audiology and Speech Sci., Purdue Univ., West Lafayette, IN 47907)

The literature on the impact of otoacoustic emissions and their underlying mechanisms on the perception of sounds will be reviewed. Although stable spontaneous otoacoustic emissions are not normally heard, any rapid change in the frequency and level of an emission may cause it to be audible. Spontaneous emissions interact with external tones which may be responsible for some instances of monaural diplacusis. Otoacoustic emissions are associated with local minima in hearing thresholds measured with small frequency increments (threshold microstructure). Changes in the level and frequency of the emissions (occurring either naturally or during aspirin consumption and noise exposure) are correlated with changes in threshold microstructure. Frequency and intensity discrimination and perceived loudness of low level tones and nonsimultaneous masking of higher level tones are modified by the presence of otoacoustic emissions.

9:40

6EA3. The effects of being a newborn on otoacoustic emissions. Susan J. Norton (Children's Hospital and Med. Ctr., 4800 Sand Point Way NE, Seattle, WA 98105 and Dept. of Otolaryngology-Head and Neck Surgery, University of Washington)

There is a growing body of data concerning otoacoustic emissions (OAEs) in newborn infants. Because OAEs are a noninvasive, objective measure of cochlear status and can be measured relatively rapidly many investigators are excited about using OAEs as a screening tool for hearing loss in newborns. Preliminary studies indicate that both transiently evoked and distortion product emissions are robust across a broad frequency range in normally hearing newborns. Also, both are absent in newborns later confirmed to have severe to profound sensorineural hearing loss. However, several questions have arisen concerning their relationship between the evoking stimulus parameters and OAE characteristics in infant versus adult ears, the influence of probe, external and middle ear acoustics on newborn responses, and the most appropriate OAE stimulus and response parameters for hearing loss screening. These issues are particularly relevant to identifying mild to moderate hearing loss and to understanding hearing development in humans. [Work supported by NIDCD.]

10:05

6EA4. Instrumentation issues in collecting ear-canal acoustic distortion products. Jont Allen (Acoust. Res. Dept., Rm. 2D553, AT&T Bell Labs, Murray Hill, NJ 07974)

Ear-canal cochlear otoacoustic emission acoustic distortion products (DPs) appear to provide a window into outer hair cell function and are becoming an important noninvasive cochlear evaluation research tool. If they are to become clinically useful, a better understanding of their production and measurement is needed. The magnitude of the DP depends on the acoustic load of the transducer as well as the cochlear input impedance. The ear-canal impedance and the ER-2 and ER-10B acoustic impedance have been measured and predictions of the reverse transfer function were made. Less well understood are observed standing waves in the ear canal at high frequencies. These may be controlled by insertion of the sound delivery tubes deep into the canal. Measurement of the DPs using an Ariel DSP-16 signal processing board is described in a portable PC-notebook configuration. DPs are approximately related to hearing level (HL) according to the formula $DP = 8 - 0.5 HL$, where both DP and HL are in dB. Several different useful measurement paradigms that have been investigated will be described.

10:30

6EA5. Transducers for otoacoustic emissions measurement. Mead C. Killion (Etymotic Res., Elk Grove Village, IL 60007 and Northwestern Univ., Evanston, IL 60208)

Most otoacoustic emissions are relatively weak as measured in the ear canal. A system noise level of -20 to -30 dB SPL is required in order to measure some emissions of interest, which means microphone noise spectrum levels of -20 dB SPL or better are needed. In the case of stimulated emissions such as the cubic distortion product at the frequency $2f_1 - f_2$, the distortion product is often more than 80 dB below the level of the f_1 and f_2 primary tones. As a result, the intermodulation distortion requirements on the amplifiers and earphones are severe—typically 0.003% or so—requiring the use of two separate earphones which must be acoustically isolated from each other. Special transducer constructions and couplings designed to meet these needs will be described, as will a new set of transducers specifically designed to be used in an inexpensive infant screening device which can be carried in the pocket like a stethoscope.

10:55

6EA6. Technique for measuring low-level otoacoustic emissions. Preeti Rao, Robert Bilger, Ann Russell, and Ted Meyer (Speech and Hear. Sci., Univ. of Illinois, 901 South 6th, Champaign, IL 61820)

To minimize background noise in the measurement of spontaneous otoacoustic emissions (SOEs), which are usually low-level signals, an effective strategy is to narrow the bandwidth of the spectral analysis. The long time required by this strategy makes it impractical in on-line measurement of SOEs, because of the effect of subject-generated noise. A system was configured that allows one to reduce the level of background noise to -13 dB SPL 8-Hz resolution. To achieve these levels, a 60-s sample of sound from a sealed ear canal (16 kHz, 16-bit resolution) was digitized and the conditioned output of a sensitive, low-noise microphone (ER-10) was stored for subsequent off-line spectral analysis. To minimize the subject-generated noise from the "quiet" recording, the resultant waveform is filtered. Digitization and analysis are achieved using an Ariel DSP-32C card. The software package used includes Hypersignal, Ariel Math-32C, and programs written by one of the authors (Rao). Examples will include (a) noise levels for the ER-10 and the ER-10B,

(b) comparison of off-line to on-line analyses, (c) fluctuations in the level and frequency of SOEs. (d) SOEs recorded from the ears of identical twins, and (e) SOEs in elderly subjects with sensorineural hearing loss. [Work supported by Grant No. DC 00174.]

Contributed Papers

11:20

6EA7. Is the "4-kHz notch" in the noise-induced-hearing-loss audiogram due to a tectorial membrane resonance at 2.8 kHz (the frequency of the maximum external ear resonance)? Mead C. Killion (Etymotic Res., Elk Grove Village, IL 60007)

If Allen is right, and the apparent transmission zero located one-half octave below the best frequency of a nerve fiber is due to the resonance of the tectorial membrane mass on its connecting stiffness, then perhaps an explanation for the half-octave shift in TTS and PTS above the stimulating frequency is available (the tectorial membrane agitation breaks off the hairs). Along the same vein, the typical diplacusis that accompanies a Meniere's attack can perhaps be explained by the conflicting periodicity pitch cues (unchanged) and place cues (shifted due to the stiffened basilar membrane). The disappearance of diplacusis over time may be an indication of the plasticity of the tonotopic organization of the brain.

11:35

6EA8. Test-retest reliability of distortion product otoacoustic emissions in normal-hearing and hearing-impaired adults. Dan C. Halling and Larry E. Humes (Dept. of Speech and Hear. Sci., Indiana Univ., Bloomington, IN 47405)

Distortion product otoacoustic emissions (DPOAEs) were measured in the ears of 25 normal-hearing and 25 hearing-impaired adults to assess test-retest reliability. Primary tones with geometric mean frequencies of 1, 2, and 4 kHz ($f_2/f_1 = 1.21$) were utilized. Results for the normal-hearing adults indicated that test and retest DPOAE detection thresholds for each ear were highly correlated at 1 kHz ($r = 0.81$), but only moderately correlated at 2 and 4 kHz ($r = 0.53$ and 0.48 , respectively). DPOAE amplitude at various suprathreshold input levels was also evaluated for test-retest reliability. DPOAE amplitudes at all input levels and frequencies showed at least a moderate correlation, with the strongest seen at 1 and 4 kHz with 75 dB input ($r = 0.85$). Of the two DPOAE measures, detection threshold and amplitude, the latter appears to be most stable over successive measurements in normal ears. The data from the hearing-impaired listeners will be used to explore further the reliability and validity of the use of DPOAEs as a clinical measure. [Work supported by NIH.]

11:50

6EA9. Why is the $2f_1 - f_2$ distortion product maximum at $f_2/f_1 \approx 1.2$? Jont Allen (Acoust. Res. Dept., Rm. 2D553, AT&T Bell Labs, Murray Hill, NJ 07974)

It has been known for a long time that the acoustic distortion product $2f_1 - f_2$ measured in the ear canal is maximum when $f_2/f_1 \approx 1.2$. It is now known that all the distortion products $f_2 - n(f_2 - f_1)$, for integer n , peak at the same frequency $f_{\max}(f_2)$, that depends only on f_2 . Neural tuning curves frequently have a notched region where the tip meets the tail. The frequency of this notch is defined as $f_z(f_{CF})$. Here, f_z depends on the characteristic frequency of the neuron f_{CF} . Assuming that the distortion products are generated on the basilar membrane at the f_2 place, f_{CF} is associated with f_2 . When this is done, it is found experimentally (for the cat) that $f_z = f_{\max}$. In other words, it has been found experimentally that the maximum of the DP is at the same frequency as where the tuning curve tail and tip meet. This result implies that the micromechanics of the cochlea contains a mechanical circuit resonance at this frequency. This impedance must be connected across the inner hair cells, so that it "shorts out" the nerve excitation, reducing the IHC sensitivity at the resonance frequency, and in series with the nonlinear generator, thereby maximally coupling the DP back into the cochlear fluid, and hence into the ear canal.

12:05

6EA10. Relaxation time constants for the suppression of spontaneous otoacoustic emissions by multiple ipsilateral tones. C. Sun, W. J. Murphy, G. R. Long, C. L. Talmadge, and A. Tubis (Depts. of Audiology and Speech Sci. and Physics, Purdue Univ., W. Lafayette, IN 47907)

Previous investigations of the time constants associated with suppression and recovery from suppression of spontaneous otoacoustic emissions (SOAEs) by pulsed external tones have shown that limit-cycle-oscillator models of SOAEs provide a basis for qualitative and quantitative interpretation of SOAE behavior [Talmadge *et al.*, *Mechanics and Biophysics of Hearing*, edited by P. Dallos *et al.*, pp. 235-242 (1990)]. The Van der Pol limit-cycle oscillator model is generalized to the case of two external driving tones, one continuous and the other pulsed. The continuous tone introduces a suppressed SOAE amplitude that is further modulated by the pulsed suppressor component. The model predicts that suppression onset and recovery times will increase with increases in the amount of suppression from the continuous suppressor tone. Preliminary data indicate that the onset and recovery times are affected in the manner implied by the model. [Work supported by NIH-NIDCD Grant No. DC 00307.]

Session 6MU

Musical Acoustics: Organ Acoustics

Herbert N. Meustadt, Chair

Electrical Engineering Department, U.S. Naval Academy, Annapolis, Maryland 21402

Chair's Introduction—8:30

Invited Papers

8:35

6MU1. Air (or water) jet as a vibrating diaphragm in aerial (or underwater) organ pipes. Shigeru Yoshikawa (5th Res. Ctr., Tech. R&D Inst., Defense Agency, 3-13-1 Nagase, Yokosuka, 239 Japan)

The jet works as not only an *energy generator* to drive the resonant pipe but also an *energy collector* to maintain its wavelike vibration over the mouth. The jet may act virtually as a flexible thin diaphragm whether it bears the jet instability or not. Its overall lateral vibration then produces the acoustic mouth current, which must correspond to the feedback current collected by the jet in the jet-drive model [J. Acoust. Soc. Jpn. (E) 1, 175–191 (1980)]. Calculation of the acoustic impedance Z'_j of the jet as a vibrating diaphragm will help one understand how the jet vibration is sustained. Although the energy dissipation that occurred in the jet is much smaller than that in the pipe, the jet as well as the pipe should maintain its own oscillation according to the proper phase relation. In other words, the velocity of the jet diaphragm should be directed out of the mouth when the acoustic pressure is positive at the mouth. The question of whether the organ pipe sounds in water as well as in air was considered theoretically from the above viewpoint and was solved experimentally [J. Acoust. Soc. Jpn. (E) 5, 211–221 (1984)]. In spite of the great difference in fluid dynamical and acoustical properties between water and air, a small underwater organ pipe made of aluminum and driven by the water jet sounded at about 155 dB *re*: 1 μ Pa near 1 kHz. Similarities and differences between aerial and underwater organ pipes will also be discussed.

9:05

6MU2. A model of jet-resonator action based on the air reed, or vibrating diaphragm, concept. S. A. Elder (Phys. Dept., U.S. Naval Academy, Annapolis, MD 21402)

Both cavity resonance, or “half-jet” drive, and organ pipe, or “full-jet” drive, can be described in terms of a model based on an extension of the “air reed” concept of Yoshikawa and Soneyoshi [J. Acoust. Soc. Jpn. (E) 1, 175–191 (1980)]. For the half-jet, or free shear layer, the effective vibrating diaphragm is the moving surface defined by the locus of points at which the shear layer velocity profile has an inflection point. Since this is the layer that separates “outside” from “inside” fluid in the shear layer, its behavior at the downstream edge determines how much external fluid is available for each cycle to drive the acoustic resonance of the cavity, considered as a parallel-resonant system (i.e., “jet drive” in control-volume type models). For the organ pipe case, the back-to-back shear layers of the full jet provide a pair of coupled “diaphragms” which act similarly. In addition to parallel-resonant action, the transverse material acceleration of the jet motion can be associated with series resonant, or “force-drive” contributions to the oscillation that arise principally from fluctuating deflection of momentum from the main jet flow. Insights from current studies on underwater half-jet resonators will be described. [Work supported by U.S. Naval Academy Research Council.]

9:35

6MU3. Transient behavior of flue organ pipes. A. Hirschberg, A. P. J. Wijnands (Eindhoven Univ. of Technology, W&S 054, P.O. Box 513, 5600 MB Eindhoven, The Netherlands), and B. Fabre (CNRS URA 868, Universite Paris VI, 4 Place Jussieu, 75212 Paris, France)

In earlier studies by Fletcher [Acustica 34, 224–233 (1976)], Nolle [J. Acoust. Soc. Am. 66, 1612–1626 (1979)], and Angster and Miklos [Proc. 13th Int. Conf. on Acoust., Vol. 3, 99–101 (1989)] crucial parameters determining the transient behavior of a flue organ pipe have been identified. In particular the jet width to mouth height ratio, the labium position relative to the jet center line, the size of the ears, and the steepness of the supply pressure rise appeared to be very important. It is interesting to relate the effect of these parameters to four fundamentally different sound production processes that occur successively in the

course of the transient: (a) the variable volume flux through the flue as a consequence of the supply pressure rise, (b) the impulsive vortex shedding occurring when the jet hits the labium for the first time, (c) the jet oscillation controlled by an edge-tone feedback mechanism (Biot-Savart induction), and (d) the jet oscillation controlled by the acoustical response of the pipe (pipe tone). After a discussion, based on flow visualization and internal pressure measurements, a simplified model is presented. The model assumes that the transient is dominated by the variable volume flux (a) and the acoustical response of the pipe (d). The jet oscillation and jet-labium interaction are described with a modified version of Fletcher's intuitive theory.

10:05-10:15

Break

10:15

6MU4. Time domain simulation and jet behavior in the flute. John W. Coltman (3319 Scathelocke Rd., Pittsburgh, PA 15235)

A time domain simulation using parameters closely imitating a Boehm flute gave mode oscillation frequencies varying properly with blowing velocity, terminating tangent to an edge-tone characteristic obtained from the same model. Second harmonic generation was much enhanced by a reflection function representing stretched modes. Stagnation pressure and hot-wire speed measurements yielded profiles for a real jet that broadened greatly under acoustic excitation, with development of a subsidiary peak. Jet swing amplitudes and phases for different frequencies agreed with pressure gradient as the stimulating force, not velocity or negative displacement. A jet blowing a flute embouchure coupled to a nonreflecting, separately excited air column, produced signals that implied an overall jet transfer function consistent with the assumptions of velocity stimulation and injected jet current drive made in the computer simulation.

10:45

6MU5. Jet displacement in edge tone, pipe oscillation, and the precursor. A. W. Nolle (Dept. of Phys., Univ. of Texas at Austin, Austin, TX 78712)

Hot-wire jet-velocity profiles are obtained at evenly spaced times throughout the steady-state acoustic cycle. Jet displacement due to an imposed acoustic crosswind at the flue exit is studied as well as that in spontaneous oscillation. The parallel-plate geometry of Coltman [J. W. Coltman, J. Acoust. Soc. Am. 60, 725-733 (1976)] is used in the edge-tone observations. Here, and also in pipe configurations, the fundamental-frequency jet displacement near the flue exit has a small phase lead relative to the transverse acoustic velocity. The acoustic radiation due to the external jet flux combines with that due to the acoustic current in the mouth in such a way as to make the external spectrum differ from that to be expected from the internal spectrum corrected for the passive acoustic transmission characteristic of the mouth; e.g., for a wide-scale stopped pipe the second harmonic amplitude is greatly enhanced in the external spectrum, and the third greatly reduced. Results for the precursor, an initial transient signal not sensitive to the pipe length, are considered briefly.

11:15

6MU6. Parametrization of organ sound. Vladimir Chaloupka (Dept. of Phys., Univ. of Washington, Seattle, WA 98195)

This paper presents first results of a research project attempting a thorough analysis of pipe-organ sounds. Sound of individual pipes, as well as of combinations of pipes, of representative organs is being systematically acquired. The analysis is performed on a NeXT computer, and includes a variety of software tools such as spectrum analysis, time-domain waveform fitting, and sinusoidal analysis *à la* McAuley-Quatieri. In the analysis and presentation of the data, the considerable number of numerical results is reduced into a compact "fingerprint" of the whole organ, enabling an easy evaluation of the tuning system, quality of the tuning, and other parameters. Interesting results are obtained on a cross-talk between simultaneously sounding pipes: it is found that under appropriate conditions, two slightly mistuned pipes of different ranks but of the same nominal pitch will lock in a single frequency when sounded simultaneously; sometimes, this common frequency is higher than either of the two original frequencies (e.g., when two pipes sounding at 525.3 and 526.4 Hz lock at 527.7 Hz). [Work supported by the University of Washington Graduate School Research Fund.]

11:45

6MU7. Musical dynamic levels of pipe organ sounds. Donald E. Hall (Dept. of Phys., California State Univ., 6000 J St., Sacramento, CA 95819-6041)

The pipe organ offers the opportunity to conduct psychoacoustic experiments in which the sound of a natural instrument can be perfectly steady and reproducible. This study took advantage of the pipe organ to concentrate on that aspect of musical dynamics determined by the physical parameters of steady sounds, leaving aside the admittedly important

effects of other variables such as context and articulation. Juries of musicians and music students provided judgments of musical dynamic levels produced by steady sounding of various stops and combinations on two pipe organs. Sound spectra from these organs have previously been analyzed in 1/3-octave bands [D. Hall, *J. Acoust. Soc. Am. Suppl.* 1 **85**, S141 (1989)]. Results of this study are compared with the hypothesis that loudness calculated by a procedure such as Zwicker's will be a good predictor of the steady aspect of musical dynamic strength, while a simple unweighted sound level in dB is rather poor.

THURSDAY MORNING, 14 MAY 1992

SALON F, 8:25 TO 11:45 A.M.

Session 6PP

Psychological and Physiological Acoustics: Binaural Hearing and Localization

Constantine Trahiotis, Chair

University of Connecticut Health Center, Surgery and Neuroscience, L4071, Farmington, Connecticut 06032

Chair's Introduction—8:25

Invited Paper

8:30

6PP1. Binaural hearing. Ervin R. Hafer (Department of Psychol., Univ. of California, Berkeley, CA 94720)

Binaural hearing was not the central focus of Professor Zwicker's long and distinguished career. However, during the 1980s he developed a strong interest in the binaural release from masking and, showing his usual fervor, he produced a long list of papers on the subject. These generally dealt with the interaction between binaural masking and some of his favorite auditory topics including: effects of such parameters as frequency, duration, and bandwidth; integration of information over time; loudness summation; recognition of vowels. Some of the currently most active topics in binaural hearing that will be discussed today include: binaural precedence and the special role of onset information in localization; the reduction of ambiguity in localization accomplished by comparing phases across frequencies in the binaural cross spectrum; localization in the open-field; generation of artificial open fields under headphones.

Contributed Papers

9:00

6PP2. The effect of a temporal notch in a diotic background on the ability to detect an interaural delay of a continuous component. M. A. Stellmack (Parnly Hear. Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

The purpose of this study was to examine the processes that produce interference when observers attempt to lateralize a target component in the presence of other components. The present study looks at the effects of a temporal notch in the diotic "distractor" components on a listener's ability to detect interaural delay of the target component. Threshold IDTs were measured for a 753-Hz tone presented against a diotic background of six additional components. The resulting seven-component complex was centered at 753 Hz with $\Delta f = 100$ Hz. During each trial,

a 753-Hz cue tone 500 ms in duration was followed by a 500-ms test interval. The distractors were gated off briefly 200 ms into the test interval and gated on again for the remainder of the test interval. The duration of the temporal notch in the distractors was 10, 20, 50, 100, or 200 ms. In separate conditions, the target component remained on for the entire test interval or was present only during the distractor notch. Threshold IDTs measured when a temporal notch in the distractors was present were approximately 80% smaller than thresholds measured when no temporal notch was present, and approached thresholds obtained for the target component in isolation. These results show that a very brief "look" at the target component in isolation is sufficient to substantially reduce the interference produced by the distractors. [Work supported by NIH.]

6PP3. The influence of an interfering band of noise on the lateral position of a target band. Lauri M. Heller (Dept. of Psychol., 3815 Walnut St., Univ. of Pennsylvania, Philadelphia, PA 19104)

Observers centered a target sound in the presence of a simultaneously gated distractor. Observers controlled either the interaural time difference (ITD) or interaural level difference (ILD) of a 50-Hz-wide target band of noise that had a center frequency of either 500 or 2000 Hz. The distractor was a 50-Hz wide band of noise with a center frequency of either 250, 500, 1000, 2000, or 4000 Hz. The target and distractor were gated on and off repeatedly within a trial, each presentation having a 150-ms steady-state portion. The ITD (or ILD) of the distractor varied from trial to trial. The effect of the distractor ITD (or ILD) on the lateral position of the target was assessed by the magnitude of the target ITD (or ILD) that was needed to center the target. The distractor ITD influenced the ITD-centered 2000-Hz target to a greater extent than the 500-Hz target, whereas the distractor ILD had an approximately equal effect on the ILD-centered targets at both frequencies. [Work supported by NIH.]

9:30

6PP4. The effects of randomizing interaural phases of components spectrally distant from the target on detection of a dichotic pitch. A. N. Grange and W. A. Yost (Parmly Hear. Inst., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

The present study examined the ability of subjects to detect an interaural phase delay of a 20-Hz-wide section of wideband noise. The narrow-band target was centered on 500 Hz and generated a dichotic, Huggins-type pitch. Thresholds for interaural phase delay detection were measured as a function of the width of the surrounding diotic noise, which was also centered on 500 Hz. Interaural phases for frequency components outside of the diotic band were drawn randomly from a rectangular distribution. The spectrum of the entire stimulus ranged from 1–1500 Hz. A two-down, one-up tracking procedure was used to estimate interaural delay thresholds, in which each trial consisted of two 500-ms noise bursts, separated by 250 ms of silence. The two intervals were identical except that the interaural delay of the 20-Hz wide target band was introduced after 250 ms in the target interval. Thresholds were obtained for diotic bandwidths of 900, 700, 500, 300, 200, 150, 125, and 100 Hz. Thresholds decreased systematically as the bandwidth of the diotic surrounding noise was increased from 100 to 500 Hz, and were unchanged as the diotic band was further widened. Comparisons were made with conditions in which the noise outside the diotic band was removed instead of being presented with random interaural phases.

9:45

6PP5. A technique for assessing spectrally analytic and synthetic binaural processing. R. H. Dye, Jr. and M. A. Stellmack (Parmly

The ability to detect interaural delay in a low-frequency target tone is hindered by the presence of diotic distractors even when target and distractors are spectrally remote. One putative source of this interference is spectral integration of binaural information arising from target and distractor components. The aim of the current study is to examine the extent to which listeners have access to information regarding which delay is associated with which spectral component. The two components were 553 and 753 Hz, with one designated as the target and the other as the distractor. The duration of the tones ranged from 50 to 400 ms, with the targets and distractors gated simultaneously. For each block of 100 trials, the target and the distractor were presented at ten different interaural delays that ranged from left leading to right leading. Each possible pairing of target and distractor delay was presented once (in a random order) during each block of trials. A cue interval, consisting of a diotic presentation of the target component, preceded each test stimulus. Subjects were instructed to indicate whether the *target* component appeared to the left or right of the intracranial midline. Left–right judgments were plotted as a joint function of the target (x axis) and distractor delays (y axis). At a duration of 400 ms, most subjects produced left–right judgments that indicated spectrally analytic binaural processing: left–right judgments were partitioned by a vertical line passing through $x = 0$. At shorter durations, left–right judgments indicated spectrally synthetic processing: left–right judgments tended to be partitioned by the diagonal $y = -x$. [Work supported by NIH.]

10:00

6PP6. Additive versus multiplicative combination of differences of interaural time and intensity. Samuel H. Tao and Richard M. Stern (Dept. of Elec. and Comput. Eng. and Biomed. Eng. Prog., Carnegie Mellon Univ., Pittsburgh, PA 15213)

Predictions of the position-variable model [R. M. Stern, Jr. and H. S. Colburn, *J. Acoust. Soc. Am.* **64**, 127–140 (1978)] are compared to the observed lateralization of bandpass noise as a joint function of differences of interaural time, intensity, and phase (ITD, IID, and IPD, respectively) [T. N. Buell and C. Trahiotis, *J. Acoust. Soc. Am.* **90**, 2266 (A) (1991)]. This presentation is primarily concerned with the effect of IIDs on the lateralization of bandpass noise. It is shown that the multiplicative intensity-weighting function that had been a feature of all previous implementations of the position-variable model is fundamentally unable to describe the data of Buell and Trahiotis. It is argued that the form of these data suggests that the effects of IID must be introduced in an additive fashion at a more central level, *after* image position is estimated on the basis of timing information alone. Such a model provides a very good description of the Buell and Trahiotis data, as well as the results of several classic studies of time-intensity trading. Also discussed are the general implications of these data in terms of peripheral and central theories of time-intensity interaction. [Work supported by NSF.]

10:15–10:30

Break

10:30

6PP7. Adaptation and release from it in free field listening. Ervin R. Hafter and Kourosh-Saberi (Dept. of Psychol., Univ. of Calif., Berkeley, CA 94720)

Using headphone presentation, it has been shown that detection of

an interaural difference of time or level in a train of n dichotic clicks is affected by the interclick interval (ICI). With ICIs > 10 ms, thresholds decline as the \sqrt{n} as predicted for optimum signal processing. With shorter ICIs, the decline is as $\sqrt{n^k}$ with $0 < k < 1$ and $k = f(\text{ICI})$. Such data have been said to show “binaural adaptation,” a process whereby each click is less effective than the one preceding. A release from bin-

aural adaption has been found in response to appropriate acoustic "triggers" such as a gap in the train or a brief burst of noise. In the present study, minimum audible angles (MAAs) were measured in the free field using two loudspeakers in the horizontal plane placed on either side of the midline. Frequency responses of the speakers were equated with digital filters. Stimuli were trains ($n = 1$ to 16) of bandpass clicks (cf. = 4 kHz) with ICIs of either 2 or 10 ms. As with headphones, shorter ICIs produced adaptation. Various potential triggers were presented in synchrony with the center of the train in order to produce a release from adaptation, and release was defined by a $\sqrt{2}$ decline in threshold relative to that for a train of length $n/2$, indicating that the $n/2$ clicks in the second half were as effective as those in the first. [Work supported by a grant from the NIDCD.]

10:45

6PP8. Discrimination of various movement patterns. David R. Perrott, Brian Constantino, Jennifer Ball, and John Cisneros (Psychoacoust. Lab., California State Univ., Los Angeles, CA 90032)

The "snapshot" hypothesis [D. W. Grantham, *J. Acoust. Soc. Am.* 79, 1939–1949 (1986)] suggests that velocity is simply inferred. Subjects are assumed to be utilizing information regarding both the distance traveled from signal onset to offset and the temporal extent of the interval during which movement occurs. A number of motion patterns were examined in the current series of experiments with particular attention to whether subjects could discriminate between sequences in which both the distances traveled and the duration of the events were identical. As a case in point, subjects were able to discriminate between acceleration and deceleration movement patterns. The implications of these results will be discussed. [Work supported by NSF.]

11:00

6PP9. Auditory systems spatial resolving power in the periphery, under serial processing, is more precise than visual spatial processing in the peripheral aspects. Brian L. Costantino, David R. Perrott, and John Cisneros (Psychoacoust. Lab., California State Univ., Los Angeles, CA 90032)

The relation between minimum audible and minimum visible angle was measured in a sequential localization task and compared. The frontal spatial fields of the visual and auditory systems are superior to their peripheral regions [Wertheim (1894); Mills (1958)]. Localization tasks were related to Vernier's acuity, except the timing was in a sequential pattern for both the visual and auditory systems. The spatial acuity of the auditory system defined space better at approximately 20-deg azimuth and beyond than the visual system. Four subjects participated,

two male and two female, ranging in ages from 23–49. Significant differences were found in the main effects of angles [$F(4,3) = 30.59, p < 0.01$] and the interaction between conditions and angle [$F(4,3) = 3.89, p < 0.05$]. The implications of these results will be discussed.

11:15

6PP10. Restrictions of computational models of monaural localization. Pierre Zakarauskas and Max S. Cynader^{a)} (Dept. of Psychol., Univ. of British Columbia, 2136 West Mall, Vancouver, BC V6T 1Y7, Canada)

In a previous contribution [P. Zakarauskas and M. S. Cynader, *J. Acoust. Soc. Am. Suppl.* 1 87, S65 (1990)], the finite-difference approximation of the first and second partial derivatives of the received spectrum with respect to frequency were investigated for their abilities to provide a sufficient basis for monaural localization. In this paper, a general description of the necessary and sufficient conditions that any transformation of the spectrum must have in order to be used for monaural localization is supplied. Both continuous and discrete transforms are considered. In particular, the Gabor function with a phase of either zero or $\pi/2$ is shown to constitute an acceptable basis for monaural localization. ^{a)}Also at Department of Ophthalmology, Univ. of British Columbia, 2550 Willow St., Vancouver, British Columbia V5Z 3N9, Canada.

11:30

6PP11. Binaural sound localization using neural networks. Rushby C. Craig (ICBM Products Directorate, Electronics Div., Hill AFB, UT 84056) and Timothy R. Anderson (Armstrong Lab., Wright-Patterson AFB, OH)

Artificial neural networks (ANNs) were used as a tool to localize sound sources from simulated, binaural signals. The sound sources for the experiments were restricted to a circle of radius 9 ft, centered about the head and lying on the horizontal circle. Sound source positions were randomly selected from one of the 360, one-deg increments on the circle. Classes for the ANNs were created by dividing the circle into equally sized wedges, much like slices of a pie. The number of classes used in the experiments varied from 4 to 36. Two types of sound source signals were considered: tones and Gaussian noise. Three different feature sets were tried. Results will be presented that compare the performance of the three feature sets for each sound source type. The best feature set produced similar results in terms of localization accuracy on tones and Gaussian noise (over 91% for 18 classes). Observations were made of phenomena which also occur in human psychological experiments such as front-back confusions and increased difficulty in localization below 1500 Hz.

Session 6SA

Structural Acoustics and Vibration: Contributed Session I

Joseph M. Cuschieri, Cochair

Department of Ocean Engineering, Center for Acoustics and Vibration, Florida Atlantic University, Boca Raton, Florida 33431

Aynur Unal, Cochair

Vibration and Sound Research Institute, 1625 Alameda, San Jose, California 95126

Contributed Papers

8:00

6SA1. Reflection from a baffled strip of a ribbed panel. G. Maidanik and J. Dickey (David Taylor Res. Ctr., Bethesda, MD 20084)

A formalism that describes the reflection properties of a ribbed panel is used to generate a formalism that describes the reflection properties of a baffled strip of a ribbed panel. The strip is defined by a pair of infinite rigid boundaries that lie normal to the direction of the parallel ribs. The formalism is demonstrated by computing the bistatic and monostatic reflections from the strip. Examples are cited for free-wave speeds that stimulate the flexural and the longitudinal free waves in a plate. In the former, the free-wave speed is dependent on the square root of the frequency and is subsonic below the critical frequency. The critical frequency is defined with respect to the speed of sound in the fluid that occupies the semi-infinite space above the panel. In the latter, the free-wave speed is supersonic and is independent of the frequency. The magnitude of the reflected pressure is presented as a function of the wave number that lies normal to the ribs and the frequency, or in some cases as a function of the monostatic angle and the frequency. The patterns exhibited in these plots are related to phenomena that are discussed.

8:15

6SA2. Higher-order theory for fluid-loaded thin elastic plates. Martin G. Manley (Graduate Program in Acoustics, Penn State Univ., P.O. Box 30, University Park, PA 16804)

An elastic, infinite plate of finite thickness with fluid loading on one side and vacuum on the other is considered. Appropriate approximations to the dispersion relation based on the full elastodynamic equations will be presented for the range $0 < kh < 2$, where k is the wave number and h is the thickness. [Supported by the PSU Applied Research Laboratory Exploratory and Foundational Research Program. The author acknowledges the advice of A.D. Pierce.]

8:30

6SA3. Vibration of finite-length, fluid-loaded ribbed plates. Richard F. Keltie (Ctr. for Sound and Vib., Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Box 7910, Raleigh, NC 27695)

The vibration response of line-forced, finite-length, fluid-loaded flat plates with attached stiffeners has been formulated. The stiffeners may be of arbitrary size and location, but for simplicity were assumed to generate transverse inertial reaction forces only. The solution for har-

monic excitation was implemented via an expansion of the *in-vacuo* plate modes. The resulting formulation required the solution of a set of simultaneous, complex linear algebraic equations for the expansion coefficients. Upon examination of the plate response at several frequencies, it was found that strong local minima in the vibration could occur at certain values of frequency. This behavior was shown to be due to a beating phenomenon between two modes. This beating arose due to the attached rib stiffeners. A quantity called the "rib function" was defined to explain this behavior. This rib function was dependent on the rib sizes, their locations, and on the *in-vacuo* mode shapes. Discussion is presented on how to use the rib function to synthesize a set of ribs to achieve a desired plate response.

8:45

6SA4. Sound radiation from finite-length, fluid-loaded ribbed plates. Richard F. Keltie (Ctr. for Sound and Vib., Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Box 7910, Raleigh, NC 27695)

The acoustic radiation from line-forced, finite-length, fluid-loaded flat plates with attached stiffeners has been formulated. The stiffeners may be of arbitrary size and location, but were assumed to generate transverse inertial reaction forces only. The solution for the plate response due to harmonic excitation was obtained using an expansion of the *in-vacuo* plate modes. The resulting acoustic radiation was obtained by numerically evaluating Rayleigh's integral. Calculated sound pressure levels were obtained in both the far field and the near field above the plate at several frequencies. Individual modal contributions to the radiated sound were also calculated. It was found that the presence of the ribs could produce local reductions in the near-field sound level by as much as 20 dB. Explanations for this behavior were found in the presence of beats between various structural modes.

9:00

6SA5. A study on source localization in plates. Yong Zhang, Anna L. Pate, and J. Adin Mann, III (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50011)

Structural intensity and a force distribution function are used to localize the sources on a plate based on the knowledge of the plate surface velocity. The plate surface velocity is measured by means of a laser vibrometer which, unlike accelerometer, does not change the plate loading and mass. Two structural intensity formulations are used. The first formulation is based on simple flexural motion that only includes

the out-of-plane motion and its special derivations. The second formulation includes all three motion components of the plate displacement. Midlin's equations of plate vibration, which take into account the rotary inertia and shear effect on flexural motion, are used to calculate the plate in-plane motions and force distribution function. The in-plane motions are then used to calculate the structural intensity. Some numerical difficulties are overcome by calculating structural intensity in k space. Results show that both structural intensity and the force distribution function can clearly show the source location.

9:15

6SA6. Vibration and acoustic response of a ribbed plate excited by a line or point force. Ten-Bin Juang, Anna L. Pate, and Alison B. Flatau (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50011)

An analytic formulation for the vibrational and acoustic responses caused by application of a point or a line force to the rib of a single-ribbed, infinite plate with negligible fluid loading has been investigated previously by the same authors ["Vibrational and Acoustic Response of a Ribbed Infinite Plate Excited by a Force Applied to the Rib," Proc. 2nd Int. Congress on Recent Developments in Air and Structure-Borne Sound and Vibration, March 1992]. An expression for the same responses of ribbed plate system due to similar excitation away from the rib was formulated. Numerical solutions were obtained by using the same technique as before. The results again show that the structural intensity in the ribbed plate is an excellent tool for force and rib localization. Acoustic radiation patterns between the previous case of force on the rib and the present case of force away from the rib are quite different as shown in 3-D pattern results. Moreover, the interaction between force and rib plays an important role in the plate's vibrational and acoustic responses.

9:30

6SA7. The vibroacoustic response of a laminated plate to a line excitation. Y. F. Hwang, P. J. Zoccola, and M. Kim (David Taylor Res. Ctr., Bethesda, MD 20084-5000)

This paper discusses a mathematical formalism for numerical calculations of the vibration and acoustic responses of a fluid-loaded laminated plate to a line excitation. The fluid loading may apply to either or both sides of the composite plate that may consist of any arbitrary number of layers of different materials. Numerical examples are presented for the acoustic radiation from the wetted surface of a soft-cored sandwich plate subjected to a line force on its dry side. Detailed analysis on the wave propagation in the layers and effects of material damping are also discussed.

9:45

6SA8. Waves in inhomogeneous axially symmetric cylindrical shells. Zhang Wang and Andrew Norris (Dept. of Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ 08855-0909)

A first-order ordinary differential equation of a displacement-rotation-stress-moment vector is derived that describes the motions of axially symmetric cylindrical shells. The shell is assumed to be homogeneous in the circumferential direction but is allowed to be inhomogeneous in the axial direction. The vector has eight components, three displacements, one rotation angle, three stresses, and one moment, and is continuous at each axial level. The equation is cast into a form that proves convenient for studying wave propagation in shells that possess inhomogeneities in the axial direction. Moreover, this equation is also equivalent to the Donnell-Yu equation. The dispersion relation and its high- and low-frequency approximations are obtained for uniform cylindrical shells that agree with earlier results. It is shown that the motions in the axial and the circumferential directions can be decoupled

from those in the radial direction at high frequencies. These motions correspond to membrane waves and flexural waves, respectively, when frequency goes to infinite. In mathematical terms, the first-order coupled differential equation can be written as two uncoupled first-order ordinary differential equations. The first one describes a four-displacement-stress vector for the axial-circumferential motions and the second one describes a four-displacement-rotation-stress-moment vector for the radial motions. The propagator matrix, the wave propagator, the transmission, and the reflection matrices are derived. [Work supported by ONR.]

10:00

6SA9. Numerical simulation of active structural-acoustic control for a fluid-loaded cylindrical shell. Chris E. Ruckman and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. & State Univ., Blacksburg, VA 24061)

Recently published studies discuss how vibrational inputs can be used to control the structural-acoustic responses of elementary structures such as beams and plates, a technique known as active structural-acoustic control (ASAC). Most of these studies use analytical models of the dynamic response. However, analytical approaches cannot be used for the practically important case of a three-dimensional structure immersed in a dense fluid, which occurs primarily in marine applications. Such fully coupled problems, in which appreciable fluid-structure interaction takes place, usually require a numerical approach. This paper describes efforts to study ASAC by computing dynamic responses of the structure/fluid system with the computer program NASHUA. Two separate feed-forward control approaches are developed and compared: a spatial-domain approach based on minimizing the radiated power, and a wave number-domain approach that does not require far-field pressure information. Singular-value decomposition aids in determining an appropriate number of control actuators and their locations. The technique is demonstrated by examining an axisymmetric, finite-length, cylindrical shell. [Work supported by ONR.]

10:15

6SA10. Extension of the surface variational principle to arbitrary motion of bodies of revolution. Jerry H. Ginsberg and Kuangcheng Wu (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

Thus far, the surface variational principle (SVP) has been implemented for curved surfaces only in axisymmetric situations. The present work begins the development of the extended principle by using complex Fourier series expansions to describe the azimuthal dependence, while the spatial variation along the shape generator is represented by a set of basis functions that depend on the parameter used in the description of the shape. When these representations are substituted into the variational quantity for SVP, integration over the surface leads to decoupling of the azimuthal harmonics. For each harmonic, the equations for the dependence along the shape generator have the appearance of the analogous axisymmetric problem, but recursion relations substantially reduce the computational effort required to evaluate the coefficients associated with the higher harmonics. The formulation is used to evaluate the surface response and far field generated by rigid body displacement and rotation of an ellipsoid and a capped cylinder. In addition, the modifications required to apply SVP to scattering problems is discussed. [Work supported by ONR, Code 1132-SM.]

10:30

6SA11. Visualization and analysis of acoustic scattering from elastic objects, on the surface and in the far field. Harry A. Schenck (Ocean Surveillance Dept., Naval Command Control and Ocean Surveillance Ctr., RDT&E Div., San Diego, CA 92152-5000)

In typical structural acoustic scattering problems, the surface pressure and normal velocity on the object are determined as well as the far-field pressure or target strength. These functions may be obtained by computation or measurement, and each may be a function of the surface coordinates as well as the direction and time or frequency characteristics of the incident wave. They are typically stored as a multidimensional array of (possibly complex) numbers. Modern visualization techniques greatly aid the analysis and understanding of these large data sets. An efficient architecture and relevant algorithms have been developed for this purpose using commercially available software and a typical workstation. A top-down approach to understanding and analyzing these data sets will be described. This includes spatial, spectral, and statistical processing of the fundamental data. The appropriate use of color, spatial coordinates of the display surface, and time animation will be illustrated with viewgraphs and videotape examples. [Work supported by the Defense Advanced Research Projects Agency (DARPA).]

10:45

6SA12. Abstract withdrawn.

11:00

6SA13. On the effects of structural joints on sound scattering. Yue-Ping Guo (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

By examining scattering from a joint connecting two perpendicularly attached plates, this paper addresses the acoustic effects of different structural joints, ranging from clamped connection, which transmits both forces (normal and tangential) and bending moment, to various pinned connections that allow free motions in one or two dimensions at the attachment point. It is shown that the normal force and the bending moment dominate the scattering process, respectively, in the low- and high-frequency domains. For low frequencies well below the plate coincidence frequency, the scattered field is dominantly given by the contribution from the normal force so that joints which support normal forces of equal value, but are otherwise different, lead to identical scattered fields. In the high-frequency region well above the coincidence frequency, the bending moment yields the dominant contribution. Thus, joints such as rotation-free hinges scatter much less sound at high frequencies than those that transmit bending moment. It is shown that the amplitude ratio of the bending moment contribution to that due to the normal force closely follows a power law of frequency (normalized by the coincidence frequency) with the power index being 1.25, which results both from the respective dipole and quadrupole characteristics of the equivalent acoustic sources due to the force and the bending moment, and from the frequency-dependent strengths of these sources that are determined by the dynamic balance of the overall coupled problem. The predominance of the normal force and the bending moment in different frequency domains is shown to also govern the energy flow into the structure. At low frequencies, this energy is in the form of longitudinal motions associated with the normal force, while at high frequencies, transverse vibrations in the structure are most energetic because most of the energy is then transmitted to the structure through the bending moment. [Work supported by ONR.]

11:15

6SA14. Negligible reflection coefficient achieved by incorporating a layer. L. Sheiba (EG&G WASC, Inc., 1396 Piccard Dr., Rockville, MD 20850), G. Maidanik, and J. Dickey (David Taylor Res. Ctr., Bethesda, MD 20084)

A configuration is modeled as a compliant layer sandwiched between two panels, with a surface impedance of which is mass controlled. The layer is placed on a basic panel that originally faced a semi-infinite fluid atop and vacuum below. The basic panel is specularly reflective

with a reflection coefficient the magnitude of which is essentially unity. Can a layer, as defined above, be designed so that, when placed atop the panel, it will cause negligible reflection coefficient at some values of (Θ, ω) ? (Here, Θ is the angle of incidence from the normal and ω is the frequency variable.) Consideration of such a design in terms of parameters that describe the properties of the basic panel, the attached layer, and the fluid are illustrated and discussed. The robustness of the design to variations in these parameters is also discussed.

11:30

6SA15. Coupling of in-plane and out-of-plane waves in thick flat plate structure subjected to a transverse load. J. M. Cuschieri (Ctr. for Acoustics and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

Using the approaches of Mindlin [J. Appl. Mech. 18, 31-38 (1951)] and Kane and Mindlin [J. Appl. Mech. 23, 277-283 (1956)] for, respectively, the in-plane and out-of-plane wave motions in thick flat plates, the equations of motion for the in-plane and out-of-plane waves are uncoupled. That is with these approaches, a transverse load will not induce in-plane motion. This is not the case if one were to use a three-dimensional elasticity solution. Also, from a physical standpoint, if a transverse load is applied on one surface of a thick flat plate structure, apart from the out-of-plane deformation, the plate will also have some in-plane deformation. The reasons behind the decoupling of the in-plane and out-of-plane waves are the simplifying assumptions of the respective approaches, where the solution is based on the midplane of the plate with symmetric or asymmetric deformation about the midplane. If a transverse load is applied on one of the plate surfaces rather than the midplane, this can be represented by two sets of load components. One set consists of two loads pointing in the same direction and represents the excitation of the out-of-plane waves and a second set that represent two "pinching" loads that induce the in-plane wave motion. Using this approach the problem of two coupled plates of different thickness is considered to determine the influence of the thickness discontinuity on the in-plane and out-of-plane waves. [Work sponsored by ONR.]

11:45

6SA16. Modal decomposition of operational response measurements in a multi-source environment. John J. Gilheany (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC 20064) and David C. Warwick (David Taylor Res. Ctr., Bethesda, MD 20084-5000)

An approach is developed combining experimental modal analysis with operational deflection measurements to determine the contributions of individual modes to the response of a structure subjected to multiple uncorrelated sources. Singular value decomposition is applied to measured cross spectral functions to decompose the response measurements into contributions associated with each significant source. A modal decomposition is then performed on fully coherent deflection patterns associated with each source. The modal participation due to each source in the total response is determined. Damping and natural frequency estimates from the modal analysis are used to determine the generalized force inputs to each mode from each source. Finally, by constraining possible degrees of freedom, estimates of the excitation force vectors associated with each source are made. To demonstrate the approach, standard experimental modal analysis techniques are applied to frequency response functions generated via a numerical model. The results are then combined with numerical simulations of both single and multi-source response measurements to obtain the modal participation spectra. These results are further reduced to determine the generalized forces and physical force vectors associated with each source.

12:00

6SA17. Resonant vibrations of fluid-loaded plates. Axisymmetric case. Christiaan Kauffmann (Faculty of Tech. Math. and

Vibrational properties of hull plates of ships are considerably modified by the presence of water. A simple model problem is presented, consisting of a thin, transversely vibrating plate with a compressible fluid at one side and clamped along a circular contour. The forced vibrations are studied by means of a Green integral representation along with a set of coupled boundary integral equations. The kernel of the integral representation is found by using the Fourier-Hankel integral transform technique and by evaluating numerically the inverse transform in the complex wave-number plane. For the axisymmetric case the boundary integral equations collapse down into a system of algebraic equations, thus yielding an exact representation for the solution. The integration along the circular contour is performed analytically by employing addition theorems for cylinder functions. The Green kernels thus obtained correspond to the response of an infinite plate subject to fluid-loading and ring-driven by constant forces and moments. Numerical results are presented that show the fluid-loading effect on the resonant frequencies of the plate, which are shifted downwards relative to the *in-vacuo* natural frequencies, while acoustic radiation contributes to the damping of the plate's resonant modes. [Work supported by the TNO Institute of Applied Physics, Delft, The Netherlands.]

6SA18. Resonant vibrations of fluid-loaded plates. Two-dimensional case. Christiaan Kauffmann (Faculty of Tech. Math. and Informatics, Delft Univ. of Technol., P.O. Box 5031, 2600 GA Delft, The Netherlands)

Vibrational properties of hull plates of ships are considerably modified by the presence of the water. A simple model problem, consisting of a thin, transversely vibrating plate with a compressible fluid at one side, is analyzed in some detail. The model equations are solved by means of a Green integral representation along with a set of coupled boundary integral equations. The kernel of the integral representation is found by using the Fourier integral transform technique and by evaluating numerically the inverse transform in the complex wave-number plane. For the two-dimensional case the boundary integral equations degenerate to a system of algebraic equations, thus yielding an exact representation of the solution. Numerical results are presented that show the fluid-loading effect on the resonant frequencies of the plate, which are shifted downwards relative to the *in-vacuo* natural frequencies, while acoustic radiation contributes to the damping of the plate's resonant modes. [Work supported by the TNO Institute of Applied Physics, Delft, The Netherlands.]

THURSDAY MORNING, 14 MAY 1992

SALON E, 8:00 A.M. TO 12:05 P.M.

Session 6SP

Speech Communication: Voicing and Vowels: Speech Production Studies and Models

Anders Löfqvist, Chair
Haskins Laboratories, 270 Crown Street, New Haven, Connecticut 06511

Chair's Introduction—8:00

Contributed Papers

8:05

6SP1. Interactive coordination of articulation and prosody in Japanese: A chronotopographic observation of laryngeal EMG. Kiyoshi Honda (ATR Auditory and Visual Perception Res. Labs., Seikacho, Soraku-gun, Kyoto, 619-02 Japan), Yukinari Yoshida (ATR Auditory and Visual Perception Res. Lab., Seikacho, Soraku-gun, Kyoto, 619-02, Japan, and Kanazawa Inst. of Technol., Kanazawa-Minami, Ishikawa, 920 Japan), and Yuki Kakita (Kanazawa Inst. of Technol., Kanazawa-Minami, Ishikawa, 920 Japan)

It is widely accepted that the cricothyroid (CT) plays a predominant role in accentual rise of F_0 while the strap muscle, particularly the sternohyoid (SH), exhibits a supplementary function in F_0 fall. Antagonistic activity has often been observed between these muscles in typical Japanese accent types. However, the activity of SH is usually less consistent with F_0 of overall utterances, mainly because the strap muscle is also involved in articulatory control for jaw opening. Therefore, the interaction should be observed between articulatory and prosody controls. This study is aimed at examining the coordination of jaw articulation and prosody production in Japanese words, using a noninvasive technique to record EMG from the CT and the strap. The signals from arrays of miniature surface electrodes attached on the neck surface are

processed to form a chronotopographic EMG, i.e., the place versus time display of neck surface electrical potential. The EMG of the CT is extracted by removing signals from the superficial muscles. The activity of the SH either to lower F_0 or to lower the jaw tends to vary with overall F_0 of utterances: the SH becomes active when the utterance is produced in low F_0 . The results suggest a complicated relationship between jaw articulation and prosody control: SH activity and the degree of jaw opening are slightly reduced when high F_0 is maintained due to sandhi, and F_0 rise tends to be suppressed at jaw lowering in some accent kernels of focused words.

8:20

6SP2. Separation of trends and subharmonic structures from random variations in fundamental frequency and amplitude. David A. Berry and Ingo R. Titze (Natl. Ctr. for Voice and Speech, Dept. of Speech Pathology and Audiology, Univ. of Iowa, Iowa City, IA 52242)

An implicit assumption in voice perturbation analysis is that one is dealing with small random variations in fundamental frequency and amplitude. However, many irregularities in voiced speech are not necessarily small or random (e.g., subharmonics, amplitude modulations,

frequency modulations, linear trends). In this study, various processing schemes are employed to detect/remove trends and subharmonics from fundamental frequency and amplitude contours. Voice perturbation measures are calculated before and after application of the detection/removal techniques. Separating trends and subharmonics from random perturbations in voice analysis may prove useful in classifying and identifying voice disorders. Results of analysis on a variety of subjects will be presented.

8:35

6SP3. Modeling a leaky glottis. Bert Cranen and Juergen Schroeter (AT&T Bell Lab., Murray Hill, NJ 07974)

During the last decade, awareness has grown that the glottis does not necessarily close completely during normal phonation. To what extent a glottal leak affects the glottal flow waveform, however, is still poorly understood. A speech production model is implemented that is capable of simulating source/tract interaction using a time-varying glottal impedance analogous to that in the two-mass glottal model of Ishizaka and Flanagan. The original formulas are extended to incorporate two different types of glottal leakage: (1) a leak that is in open connection with the membranous part of the glottis, and (2) a leak formed by a separate duct (e.g., in the posterior commissure). However, instead of letting the glottal geometry be defined by a self-oscillating mechanical model of the vocal folds, the glottal inlet and outlet areas are prescribed by the authors. In the present simulations it is found that for a moderate leak area the flow through a type 2 leak and the flow through the membranous part may be in opposite directions during the interval that the folds close, thus enhancing the excitation of the vocal tract. Another interesting consequence of this result is that a physical explanation might be given for some of the waveform details sometimes observed in inverse filtered speech that are often considered as artifacts of the inverse filtering method.

8:50

6SP4. Effects of RLN and SLN stimulation on glottic area. Steven A. Bielamowicz, Gerald S. Berke, Deborah Watson, and Ye Ming (Div. of Head and Neck Surgery, UCLA Med. Ctr., Los Angeles, CA 90024)

In vivo canine experiments have demonstrated that vocal fold stiffness varies proportionately with changing levels of recurrent laryngeal nerve (RLN) and superior laryngeal nerve (SLN) stimulation. This study evaluates the morphologic changes in the glottis at varying levels of nerve stimulation and the effects on laryngeal resistance. Stroboscopic data from the *in vivo* canine model were examined under varying conditions of RLN and SLN stimulation. Computerized analysis of stroboscopic images was used to reconstruct the glottic area versus time waveforms and the results will be presented. Glottal dynamics were measured using photoglottography, electroglottography, and the glottal area waveforms. The effects of varying RLN and SLN stimulation on laryngeal resistance will be discussed. [Work supported by an American Laryngologic Research Award.]

9:05

6SP5. Measurements of mucosal wave propagation and vertical phase difference in vocal fold vibration. Ingo R. Titze (Dept. of Speech Pathology and Natl. Ctr. for Voice and Speech, Univ. of Iowa, Iowa City, IA 52242), Jack J. Jiang (Dept. of Otolaryngol., Northwestern Univ. School of Medicine, Chicago, IL 60611-2232), and Tzu-Yu Hsiao (Dept. of Otolaryngol., Natl. Taiwan Univ. Hospital, Taipei, Taiwan)

Examination of surface wave properties in the vocal fold mucosa is becoming an important part of assessment of vocal function. A key wave property is propagation velocity, which determines the phase de-

lay between upper and lower margins of the vocal folds. Excised canine larynges were used in this study to measure this phase delay, and therefore propagation velocity. The motion of two fleshpoints was tracked stroboscopically. Differential displacements between the fleshpoints were matched to displacements of a model. A least-squared fit of the data to the model provided the numerical values of propagation velocity, which varied from 0.5 m/s to about 2.0 m/s, depending on fundamental frequency. The corresponding phase delay along the medial surface of the vocal folds varied from about 60°/mm to 30°/mm. [Work supported by NIDCD, Grant No. P60-DC0976.]

9:20

6SP6. Acoustic analysis of glottal gap measurement. Jong-Daw Yu, Kazunori Mori, and Stanley M. Blaugrund (Ames Vocal Dynamics Lab., 100 East 77th St., New York, NY 10021)

A simple, efficient, noninvasive method for defining the size of the glottic gap in patients with dysphonia would be highly desirable. Current methods for defining the degree of insufficiency involve aerodynamic and videostroboscopic analysis. PARCOR analysis of the glottal area was calculated utilizing DAT recorded data obtained at a 44.1-kHz sampling rate from 15 patients with dysphonia, 14 of which presented unilateral recurrent laryngeal nerve paralysis. Ten thousand data points were calculated for each voiced sample. PARCOR coefficients were obtained and used as a basis for determining the cross-section area of vocal tract, i.e., the glottal area ratio (GAR). GAR, which is a noninvasive technique, was utilized successfully in the verification of the reduction in glottal gap of 15 patients who underwent medialization thyroplasty for glottis insufficiency. Results demonstrate that GAR utilizing PARCOR analysis corresponds favorably with findings obtained using well-attested methods of aerodynamic and videolaryngoscopic analysis.

9:35

6SP7. On the existence of speaker-specific maximum flow declination rate (MFDR)-sound pressure level (SPL) profiles. Arend M. Sulter and Harm K. Schutte (Voice Res. Lab., Dept. of Otorhinolaryngol., Univ. Hospital Groningen, P.O. Box 30.001, 9700 RB Groningen, The Netherlands)

Oral flow was registered with a Rothenberg mask in two male subjects during the utterance of an /æ/ vowel embedded in a word at three intensity levels (soft, normal, and loud voice), and during the production of crescendos and decrescendos on the same vowel at three fundamental frequencies. Registered signals were digitized and processed with specific hard- and software [J. S. Perkell, *J. Acoust. Soc. Am.* **89**, 1777-1781 (1991)] to give the MFDR. This is a measure for the decrease of airflow through the glottis, during the closing phase of the glottal cycle. MFDR values showed a speaker-specific relationship with SPL. To explore this specificity, MFDR values were measured in 70 subjects during phonation of the vowel /æ/ with soft, normal, and loud voice. The relation between MFDR and SPL can be described with $MFDR = a + b^{SPL}$. The dependency of the parameters *a* and *b* on gender, voice training, and vocal pathology will be discussed. [Support was given by the Foundation for Linguistic Research, NWO.]

9:50

6SP8. Using beam elements to model the vocal fold length in breathy voicing. Mark A. Johnson (Res. Lab. of Electron. and Dept. of Electrical Eng. and Comput. Sci., MIT, Cambridge, MA 02139)

During breathy voicing, the vocal folds close in a zipperlike fashion from front to back, rather than closing all at once as they do during modal voicing [D. H. Klatt and L. C. Klatt, *J. Acoust. Soc. Am.* **87**, 820-857 (1990)]. The two-mass model of fold vibration can only simulate this nonuniform closure if it is modified to allow nonuniform

displacement along the length of the folds. A simple model is obtained if the rigid masses of the two-mass model are replaced by three-node Mindlin beam elements, with the anterior and posterior ends of each beam pinned. During modal voicing, the two-beam model is almost identical to the two-mass model. With the arytenoids separated for breathy voicing, the two-beam model behaves like the two-mass model,

until the folds start to close. As the beam elements close from front to back, the shear stiffness of the open part increases relative to its mass, so that the back part never quite closes. The combination of gradual closure and posterior ducting gives the spectral tilt characteristic of breathy voicing. Waveforms and spectra from preliminary simulation will be presented. [Research supported in part by an NIH grant.]

10:05-10:20

Break

10:20

6SP9. Aerodynamics of phonation: Effect of consonantal context on vowels. Richard S. McGowan, Laura L. Koenig, and Anders Löfqvist (Haskins Lab., 270 Crown St., New Haven, CT 06511-6695)

Recordings of oral airflow have been made of four subjects producing VCV sequences with different middle consonants and an open vowel. The flow signals at vowel offset and onset have been inverse filtered using a software inverse filter and characterized as a function of time using measures of maximum flow, minimum flow, open quotient, and area of the glottal pulse. Results suggest that the aerodynamic properties of the source differ considerably among vowels following different consonants. For example, at the onset of a vowel following a voiceless stop or fricative, the source typically shows a breathy pattern of vibration, as evidenced by high values of peak flow and an open quotient close to 1; the source then changes gradually into a modal type of phonation. These variations can be explained by different observed patterns of coordination between the larynx and the oral articulators. The present study reports experimental data and simulations of such dynamic patterns of source variations using a two-mass model of the vocal folds incorporated into a low-frequency model of the vocal tract's aerodynamics. [Work supported by NIH.]

10:35

6SP10. Patterns of correlation of tongue movement and electromyographic signal in vowel production. Katherine S. Harris (Graduate School, City Univ. of New York, 33 W. 43rd St., New York, NY 10036-8099), Eric Bateson (ATR Res. Lab., Kyoto, Japan), and Peter Alfonso (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

X-ray microbeam data from pellets on the jaw, tongue, and lips were compared with EMG signals from eight muscles for production of multiple tokens of 11 vowels in /əpVp/ context, for an American English speaker. Pellet data were obtained from the Tokyo x-ray microbeam while electromyographic data were obtained at Haskins Laboratories. Acoustic recordings and a comparison of orbicularis oris electromyographic records with lip pellet movement data were used to link the two data sets, so that an estimate of muscle contraction time could be made and appropriate correlations of time functions across the sets generated. Tongue pellet data were adjusted to compensate for the effects of jaw position. Relations between EMG and movement signals for the point vowels /i/, /a/, and /u/ were as expected, in that movements up and front were accompanied by large signals from anterior and posterior genioglossus, movements up and back were accompanied by large signals from styloglossus, while tongue lowering movements were accompanied by substantial hyoglossus signals. However, relations among muscle signals were unique to each vowel. [Work supported by NIDCD grants to Haskins Laboratories.]

10:50

6SP11. Forward-reverse evaluations of American English monophthongs. James D. Miller, D. J. Meyer, and Michael Gottfried (Central Inst. for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Members of a corpus of 2304 vowels [Fourakis, J. Acoust. Soc. Am. **90**, 1816-1827 (1991)] were assessed for steady-state and dynamic spectral bases for their perceptual identities. Each putative steady-state segment as well as the entire vocalic nucleus was studied. Four listeners participated in a series of perceptual experiments. All sounds were identified as one of nine monophthongs. Only correctly identified tokens were studied in a "forward-reverse rating task" and a "forward-reverse identification task." In the rating task, the listeners judged the phonetic similarity of forward and backward playings of each token. In the identification task they identified forward and backward playings as one of 16 monophthongs and diphthongs. Stimuli reliably and correctly identified whether played forward or backward are treated as "steady state." Stimuli reliably and correctly identified when played forward, but not so identified when played backward, are treated as "asymmetrical spectral glides" or, similarly, as exhibiting "vowel inherent spectral change" [Nearey and Assmann, J. Acoust. Soc. Am. **80**, 1297-1308 (1986)]. Classification schemes are discussed. [Work supported by NIDCD.]

11:05

6SP12. Acoustic correlates of American English diphthongs. Michael Gottfried, James D. Miller, and D. J. Meyer (Central Inst. for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Various studies on diphthongs suggest that they can be effectively classified in terms of either (1) the pattern of the fundamental frequency and formants at the onset and offset of the production, (2) the onset pattern and the F_2 rate of transition, or (3) the onset pattern and the direction of formant movement in an acoustic space [Nearey and Assmann, J. Acoust. Soc. Am. **80**, 1297-1308 (1986)]. Extending earlier reports [Gottfried, J. Acoust. Soc. Am. **89**, 1997 (1991)] these three hypotheses were assessed in (a) a $\log F_1 \times \log F_2$ space and (b) an auditory-perceptual space. Values for the relevant parameters were obtained for a corpus of 768 productions of American English diphthongs (6 diphthongs \times 4 speakers \times 2 tempos \times 2 stress conditions \times 2 consonantal contexts). The hypotheses were then evaluated in terms of classification performance using a statistical pattern recognition procedure. Results indicate that the acoustic parameters specified in each hypothesis were effective in obtaining correct classification of diphthongs. [Work was supported by NIDCD.]

11:20

6SP13. A cross-language study of vowel production and perception: A pilot study. Ann R. Bradlow (Dept. of Modern Languages and Linguistics, Morrill Hall, Cornell Univ., Ithaca, NY 14853)

This paper presents the results of a pilot experiment for a larger

study of the effect of inventory size and structure on vowel production and perception. This study will use acoustic and perceptual data from languages which vary in the size and structure of their vowel inventories to address questions regarding the connection between vowel production and perception, and regarding the universality of language specificity of vowel space categorizations. The data for this pilot study are from four speakers of each of two languages (English and Spanish), which differ in the size of their vowel inventories: English has 11 distinct vowel qualities whereas Spanish has 5. For the production study, recordings of words which illustrate the vowel contrasts were digitized and the formant frequencies of the target vowel steady states were measured from LPC spectra. The resultant vowel charts were plotted using various scales (koenig, mel, bark, log frequency), which were then compared in terms of their efficiency in representing the distinct acoustic categories. A perceptual experiment is planned in which listeners will be asked to categorize synthetic stimuli which will be constructed on the basis of the acoustic parameters measured in the production study.

11:35

6SP14. The phonemic transformation effect: Intersubject agreement on verbal forms. Magdalene H. Chalikia (Dept. of Psychol., Moorhead State Univ., Moorhead, MN 56563), Richard M. Warren, and James A. Bashford, Jr. (Dept. of Psychol., Univ. of Wisconsin—Milwaukee, Milwaukee, WI 53201)

Earlier studies found that repeated sequences of brief steady-state vowels are heard as verbal forms consisting of syllables occurring in English. The present investigation employed six different arrangements of four 60-ms vowels. Unlike earlier studies, a 300-ms silent gap separated two of the items, so that verbal organization of different subjects started at the same stimulus position. Individual listeners reported their verbal organizations corresponding to each arrangement. Then, listeners were assigned to a partner, and each member of the pair attempted to

match the verbal organization reported by the other to the appropriate sequence. Most listeners achieved a perfect correspondence of verbal forms with sequences for each of the six stimuli. The procedure was repeated substituting whispered for voiced vowels, and similar results were obtained. It appears that verbal organization of vowel sequences is not completely idiosyncratic, but is based upon objective acoustic characteristics resembling specific English syllables. [Work supported by NIH Grant. No. DC00208.]

11:50

6SP15. Estimation of the acoustical transfer function from vowel confusions. Matthias Wesselkamp and Birger Kollmeier (Drittes Physik. Inst., Univ. Göttingen, Bürgerstr. 42-44, W-3400 Göttingen, Germany)

An attempt is made to estimate a spectral alteration in the acoustical transfer function (including the transmission characteristic of the peripheral auditory system) from vowel confusions. To study the relation between a predetermined spectral alteration and the resulting vowel confusions in a closed nonsense-syllable test, different conditions of filtered speech stimuli were employed with and without interfering speech-spectrum-shaped noise. The number of available perceptual cues and the variability of the material was restricted by synthesizing speech with a LAR vocoder. The vowel confusions between 14 syllables were analyzed with the transinformation analysis for the different filtered conditions. For this purpose, a set of 15 acoustical features was constructed. Each feature represents the average log power within one out of 15 auditory critical bands. Using these features, the "transformation spectrum" is computed, i.e., the transmitted speech information as a function of frequency. This quantity might be used to predict the acoustical transfer function. Based on the empirical results, the correspondence between alterations in the acoustical transfer function and changes in the transinformation spectrum will be discussed.

THURSDAY MORNING, 14 MAY 1992

SALON D, 7:55 TO 11:45 A.M.

Session 6UW

Underwater Acoustics: Stochastic Acoustic Modeling

Terry E. Ewart, Chair

Applied Physics Laboratory, University of Washington, 1013 East 40th Street, Seattle, Washington 98195

Chair's Introduction—7:55

Invited Papers

8:00

6UW1. Stochastic matched-field processing. Arthur B. Baggeroer (MIT, Cambridge, MA 02139) and William A. Kuperman (Naval Res. Lab., Washington, DC 20235)

Matched-field processing (MFP) exploits the acoustic field to estimate either a source position or environmental parameters. Many have noted that the performance of MFP degrades rapidly with uncertainty about the field especially if adaptive methods are used to suppress sidelobes. It is clear that robust MFP methods are needed for working with experimental data. There are three categories of uncertainty that have been examined: (i) observational (array geometries and sensor responses are imprecisely known), (ii) statistical (ambient field covariances have errors), and (iii) environmental (the propagation is uncertain because of oceanographic characterization). The third category is examined in this presentation. The fundamental issue for robust MFP is to match the stochastic propagation with array processing that responds to an ensemble of replicas not just one from a deterministic model. At high SNRs one can search

the ensemble for the appropriate sample vector using efficient searching algorithms such as simulated annealing. At modest SNRs the processing must incorporate errors from the noise. The approaches to MFP with environmental uncertainty are reviewed and some of the relevant signal processing literature for the "detection of random signals in noise" are noted. [Work supported by Mathematical Sciences, ONR.]

8:25

6UW2. The parabolic moment equations—Has their promise been fulfilled? B. J. Uscinski (Dept. of Appl. Math. and Theor. Phys., Univ. of Cambridge, Silver St., Cambridge CB3 9EW, England)

The parabolic equations for moments of a wave field in a weak random medium are attractively simple. They are linear second-order partial differential equations of the diffusion equation type. They involve the random medium only in terms of its two-point correlation, and most importantly they are mutually independent, so no closure condition is needed. In practice these equations have proved very difficult to solve. Despite the very solid results that have been obtained there are almost as many unresolved difficulties. A full solution has been found for the second moment at a single frequency, but it cannot naturally include the case of a curving ray path in anisotropic scattering irregularities. The cross-frequency second moment describes pulse spread, but closed solutions exist only for certain types of correlation functions. Solutions for the fourth moment tend to break down in the cross-frequency case. Attempts to solve equations for the higher moments have so far failed completely, leaving the question of the probability distribution still open. Despite these difficulties, however, the parabolic moment equations still offer one of the best vehicles for exploring wave propagation in random media.

8:50

6UW3. Stochastic modeling of bottom scattering. Darrell R. Jackson (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

Several basic issues arise in stochastic modeling of bottom scattering. Most models predict the mean intensity (a second moment) in terms of an interface scattering cross section per unit area. The cross section is defined under asymptotic conditions, e.g., the transmitter and receiver are infinitely distant, the propagation medium is homogeneous, and time-harmonic excitation is assumed. The key question is the following: When is the cross section thus defined applicable to realistic conditions? Even when the cross section is of no use, the second moment may still have utility, particularly if the scattered field obeys Gaussian statistics. Data are available at high frequencies showing both Gaussian and non-Gaussian behavior of the backscattered field [Chotiros *et al.*, *J. Acoust. Soc. Am.* 77, 975–982 (1985)]. In the case of low-frequency bottom scattering, the division of bottom topography into large scales (to be treated deterministically) and small scales (to be treated stochastically) is an important issue. Here one must consider acoustic, geological, and practical criteria in making the division.

9:15

6UW4. Models of the distribution of received power in a random propagation medium. James H. Churnside (NOAA Wave Propagation Lab., 325 Broadway, Boulder, CO 80303)

The probability density function of the received power of an acoustic signal that has propagated through random sound speed fluctuations in the ocean can be obtained exactly only for two cases. The first case is where the power fluctuations are very small (much less than the mean value). In this case, the density function is lognormal. The second case, which is not realizable, is that of extremely large sound speed fluctuations or path lengths. This condition would produce a negative exponential density function of acoustic power fluctuations. A number of models for the probability density function of received power have been suggested for the region in between these two extremes. A successful model should be possible to evaluate numerically, should be physically plausible, and should describe available data. Candidate density functions will be reviewed, with an emphasis on those that were originally developed to describe optical propagation through the atmosphere.

9:40

6UW5. A partial review of path integrals for waves in random media. David H. Berman (Dept. of Phys. and Astron., Univ. of Iowa, Iowa City, IA 52242)

Path integrals are a convenient way to express exact solutions of the parabolic equation for wave propagation in variable media. Although path integrals can rarely be evaluated in closed form, the dependence of path integrals on spatially varying indices of refraction is explicit. This makes path integral solutions of the wave equation well suited for describing effects of randomly fluctuating refractive indices, and also well suited to developing classes of approximations that might otherwise go unnoticed. Moments of fluctuating fields are formed by approximation of exact path integral solutions. This is in contrast to moment methods, in which exact differential equations for the moments are approximated and then solu-

tions for these approximate equations are developed. Here approximations for the path integral which replace integration over all paths by integration over a restricted class of paths will be discussed. This procedure gives a variant of the Rytov approximation, and moments are computed as if there were an effective phase screen. In appropriate limits this variant of the Rytov approximation gives Born scattering and the standard eikonal or WKB approximation. [Work supported by U.S. Naval Research Laboratory.]

10:05-10:15

Break

Contributed Papers

10:15

6UW6. Testing the results of stochastic modeling. T. Ewart and S. Reynolds (Univ. of Washington, Mail Stop HN-10, Seattle, WA 98105)

It is essential that models of the stochastic behavior of a propagating wave field represent experience. In the field of wave propagation in random media, few data sets are available where the stochastic index of refraction fluctuations are known sufficiently well to allow their use as input to stochastic propagation models. Only in those few cases can the model predictions be compared with the observed wave statistics. Interestingly enough, the statistical stability of ocean internal waves makes the study of this phenomena in the context of ocean acoustics a mechanism for model testing. Here a few of the available data sets are discussed and the success of the testing process is assessed. Data sets may be obtained using Monte Carlo methods to generate the random medium. Various propagation models then produce the statistically varying random fields. Such methods are valuable when used to provide initial testing of models, and for extrapolation of the results to ranges of scattering parameters not available in field experiments. Examples comparing predicted and measured moments of a propagating wave field will be discussed. A summary of unanswered questions will be proposed.

10:30

6UW7. Parabolic equation average propagation routine: Some results. Roger M. Oba (Numerical Modeling Div., Code 221, Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Results from a routine for computing the average solution to the generalized parabolic equation are presented. The parabolic equation average propagation routine (PEAP) uses profiles that are depth dependent but piecewise range independent with common intervals of invariance [R. M. Oba, *J. Acoust. Soc. Am.* **90**, 2300 (1991)]. It uses analytically computed averages of normal mode solutions of the parabolic wave equation. Its application in the case that the probability distribution can be parametrized by one variable are demonstrated. Validation against averages of multiple runs of deterministic models have been performed. Features of the average solution transmission loss and phase change are shown and compared to deterministic solutions. The extensions to several parameters will be discussed. [Work supported by ONR/NRL-SSC and ONR Young Navy Scientist Program.]

10:45

6UW8. Incoherent ocean bottom scattering from non-Gaussian slopes in the Pacific. Ronald L. Dicus (Sci. Appl. Intl. Corp., 1710 Goodridge Dr., McLean, VA 22102)

Kirchhoff scattering theory was applied to measurements of bottom scatter to explain the time and angle spread of signals received on a horizontal array. The data were collected during "Pacific Echo II," a joint experiment conducted by the U.S. Naval Research Laboratory and

the Canadian Defence Research Establishment Pacific. Source signals were standard MK-61 SUS charges detonated at 244-m depth. The array was 300 m long, towed at 4 kn and 200-m depth. The 64 hydrophone groups of the array were equally spaced. The site was characterized by an average sediment thickness of 25 m. Water depth was 5000 m permitting time separation of arrivals having different numbers of bottom bounces. A slope density function was constructed partially from SEABEAM measurements at the site and partially from documented DEEP TOW measurements at other Pacific thin sediment sites. Predictions of omni- and beam-formed time series and an angle spread function were calculated and compared with the data. Results show that the observed long coda and large angle spreads may be predicted by the long non-Gaussian tails of the slope density function.

11:00

6UW9. Bispectral analysis of ocean volume reverberation. Dimitri Alexandrou and Vincent Premus (Dept. of Elec. Eng., Duke Univ., Durham, NC 27706)

The univariate probability density function (pdf) of the volume reverberation process is typically modeled as symmetric and non-Gaussian, with a variable kurtosis parameter. However, very little is known about the joint distribution of volume reverberation. In this study, the bispectral analysis of actual ocean volume reverberation data is performed in an effort to gain more insight into the structure of the joint probability density function of the volume reverberation process. The test vehicle is the Hinich test, a statistical method based on the sample bispectrum, which can be used to quantify the degree of non-Gaussianity and asymmetry in the joint pdf of a random process. Simulated data records, distributed χ^2 with decreasing numbers of degrees of freedom, are first examined to provide a baseline measure of test statistic behavior as a function of pdf asymmetry. The test is then applied to actual volume reverberation data records obtained using SEA BEAM. Although the empirically determined *univariate* density function for each SEABEAM data record exhibits a very high degree of symmetry, the Hinich test statistic indicates the presence of a large degree of asymmetry in the *joint* pdf of the volume reverberation process.

11:15

6UW10. Effect of randomly inhomogeneous seafloor on the reflection and scattering of plane acoustic waves in shallow waters. Mohsen Badiey, Alexander H.-D. Cheng, Indra Jaya, Susan McGeary (College of Marine Studies, Univ. of Delaware, Newark, DE 19716), and Morris Schulkin (Ocean Acoust., Inc., Potomac, MD 20854)

The bottom of the ocean is composed of randomly inhomogeneous, anisotropic layers of sediments. A geoacoustic data set is constructed from previously measured geological experiments in shallow-water regions of the New Jersey Continental Shelf. This data set is used as input for a stochastic reflection model based on the Monte Carlo technique in order to study the effect of the random variations of the seafloor prop-

erties on the scattering and reflection of acoustic plane waves in a shallow ocean. It is found that the random inhomogeneities as well as the anisotropy of the subsurface are major causes of volume scattering and certain anomalies in the reflection coefficient.

11:30

6UW11. Source localization in an uncertain scattering environment.
G. Haralabus, V. Premus, D. Alexandrou, L. W. Nolte, and A. M. Richardson (Dept. of Elec. Eng., Duke Univ., Durham, NC 27706)

The sensitivity of conventional matched-field processing algorithms to uncertainty in environmental parameters has prompted the design of more robust methods for source localization. Recently, the development of a new algorithm, the optimum uncertain field processor (OUPF),

was reported which incorporates, *a priori*, the environmental uncertainty into the design of the matched-field processing algorithm [A. M. Richardson and L. W. Nolte, *J. Acoust. Soc. Am.* **89**, 2280–2284 (1991)]. The processor was demonstrated to successfully locate a source in a deep ocean environment for the case of an uncertain sound velocity profile. The present study addresses the problem of source localization in a shallow-water channel, in which the received pressure field consists primarily of bottom and surface interacting rays. Surface and bottom roughness are parametrized by the Eckart reflection coefficient. Environmental uncertainty is then introduced through mismatch in the reflection coefficient. Initial simulation results demonstrate that, when scattered rays constitute a significant portion of the received pressure field, the OUPF outperforms the conventional matched-field processor when the actual reflection coefficient is imperfectly known.

THURSDAY MORNING, 14 MAY 1992

SALON H, 9:30 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

EG & G Mason Research Institute, P.O. Box 9024, Albuquerque, New Mexico 87119-9024

H. E. von Gierke, Chairman, respective U.S. Technical Advisory Groups (TAGs) for ISO/TC 43/SC1 and ISO/TC 94/SC12

1325 Meadow Lane, Yellow Springs, Ohio 45387

Standards Committee S12 on Noise. Working group chairs will report on their progress under the plan 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 Chair of the respective U.S. Technical Advisory Groups (H. E. von Gierke) will report on current activities of these International Technical Subcommittees under ISO. A report will be given on the last meeting of ISO/TC 43/SC1, which took place in Australia, from 5–12 December 1991.

Session 7AO

Acoustical Oceanography: Acoustical Determinations of Ocean Parameters and Processes

Michael J. Buckingham, Chair

Marine Physical Laboratory, Scripps Institution of Oceanography, La Jolla, California 92093-0213

Chair's Introduction—1:30

Contributed Papers

1:45

7AO1. Seismic conversion of low-frequency surface-generated ambient noise in the deep ocean. H. Schmidt, J.-Y. Liu (MIT, Cambridge, MA 02139), and W. A. Kuperman (Naval Res. Lab., Washington, DC 20375)

The strong increase in ambient noise observed at low frequencies (1–10 Hz) in shallow water can be explained by the strong excitation of seismic interface or Scholte waves by surface sources in this frequency regime [H. Schmidt and W. A. Kuperman, *J. Acoust. Soc. Am.* **84**, 2153–2162 (1988)]. There is strong experimental evidence that a similar increase observed in deep water is also due in part to the excitation of Scholte waves [Dorman *et al.*, *J. Acoust. Soc. Am.* **89**, 1905–1906 (1991)]. However, here the separation of the surface and bottom is too large to allow for direct coupling of surface-generated noise into the evanescent seismic waves. Here, the hypothesis that the Scholte waves are excited through rough interface scattering of the surface generated noise is investigated. By combining the Schmidt–Kuperman noise model with a perturbation formulation for three-dimensional scattering by rough interfaces in the bottom [W. A. Kuperman and H. Schmidt, *J. Acoust. Soc. Am.* **86**, 1511–1522 (1989)], it is demonstrated that rough interface scattering can account for a significant part of the observed increase in noise level at the seabed. In addition it is shown that the spatial coherence predicted by this model is consistent with the experimental observations. [Work supported by ONR and NOARL.]

2:00

7AO2. Very low-frequency seismo-acoustic noise below the seafloor. C. R. Bradley and R. A. Stephen (Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

The low-frequency acoustic seismic experiment (LFASE), conducted in the Blake Bahama Basin in August and September of 1989, was the first experiment to measure VLF ambient ocean noise on a seismic array within a DSDP borehole. Four three-component geophones were clamped to the borehole wall at 10, 40, 70, and 100 m below the seafloor. The frequency response of the acquisition system was flat from 4.5 to 40 Hz but meaningful ambient noise measurements were acquired down to 0.3 Hz. Both discrete sources (interpreted as local microearthquakes) and a continuous distributed source (identified as a mid-Atlantic storm) were observed. The earthquakes are characterized as sudden arrivals of high-amplitude (at least 10 dB above background), high-frequency (above 20 Hz) signals with a coda decaying to background levels within 30 s. Within the upper 10 m of the seafloor these events have dominantly horizontal motion. Vertical motion becomes dominant at 70 m and by 100 m the events are indistinguishable from background noise. These amplitude effects with depth in the array suggest that these are Stoneley waves trapped near the seafloor. Noise

levels associated with the continuous source (storm) also decay significantly below the sediment/water interface. The noise levels associated with the storm appeared below 2 Hz. Between 0.3 and 1.0 Hz there is a strong directionally coherent signal. Analysis of particle motions indicates that the source is in the direction of Hurricane Erin, which was located off the West African and European coast. Below 1.0 Hz the particle motion of the storm-related energy is clearly retrograde elliptical at the seafloor indicating Stoneley wave modes. [Work supported by ONR.]

2:15

7AO3. Predictions and observations of seafloor noise generated by sea surface orbital motion. Timothy E. Lindstrom and George V. Frisk (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

A model is developed for the prediction of the seismo-acoustic noise spectrum in the microseism peak region (0.1 to 0.7 Hz). The model uses a theory developed by Cato [*J. Acoust. Soc. Am.* **89**, 1096–1112 (1991)] for an infinite depth ocean in which the surface orbital motion caused by gravity waves may produce acoustic waves at twice the gravity wave frequency. Using directional wave spectra as inputs, acoustic source levels are computed and incorporated into a more realistic environment consisting of a horizontally stratified ocean with an elastic bottom. Noise predictions are made using directional wave spectra obtained from the SWADE surface buoys moored off the coast of Virginia and the SAFARI sound propagation code, with a bottom model derived using wave speeds measured in the EDGE deep seismic reflection survey. The predictions are analyzed for noise level variations with frequency, wave height, wind direction, and receiver depth. These predictions are compared to noise measurements made in ECONOMEX using near-bottom receivers located close to the surface buoys. Good agreement is found between the predictions and observations under a variety of environmental conditions. [Work supported by ONR.]

2:30

7AO4. Long-term 0.05–5 Hz ambient deep-ocean noise, wind, and ocean waves. Charles S. McCreery, Frederick K. Duennebie, Daniel A. Walker, and Thomas A. Schroeder (School of Ocean and Earth Sci. and Technol., Univ. of Hawaii, 2525 Correa Rd., Honolulu, HI 96822)

Long-term ambient ocean noise data from the Wake Island hydrophone array in the northwest Pacific are compared to wind measured at Wake and to ocean wave estimates near Wake from U.S. Navy models. Between 0.05 and 0.1 Hz, the hydrophone data are limited by system noise, although long-period signals from moderate to large earthquakes often exceed that limit. From 0.1 to 0.2 Hz, the ambient noise correlates

well with the estimated long-period ocean swell, with about a 2.5:1 correspondence in their respective frequencies—close to the 2:1 correspondence predicted by nonlinear wave interaction theory. Between 0.2 and 0.3 Hz, the most energetic part of the noise spectrum, the correlation with ocean waves is weaker, suggesting that this noise may have its origin at a more distant location. Between 0.3 and 1 Hz, the noise again correlates strongly with the estimated ocean waves, although the frequency correspondence is 5:1 or greater. And from 1 to 5 Hz, the ambient noise correlates very strongly with windspeed and exhibits a saturation that probably corresponds to the saturation of short wavelength ocean wind waves. Based on hourly measurements made over a 4-yr period, ambient noise levels between 0.1 and 2 Hz vary by less than 20 dB, 90% of the time. Between 2 and 5 Hz, noise levels are found to be saturated more than 70% of the time.

2:45

7A05. Space-time characteristics of broadband seismic noise: Results of the onshore component of SAMSON. John Nábělek, Guibiao Lin, and Anne Tréhu (College of Oceanography, Oregon State Univ., Corvallis, OR 97331)

In Oct. and Nov., 1990, data from an array of 37 three-component seismometers (16 broadband, 6 5 s, 15 1 s) deployed on the coastal plain of northeastern North Carolina and southeastern Virginia were collected as a part of the multidisciplinary SAMSON (Sources of Ambient MicroSeismic Oceanic Noise) experiment. The array extended from a few hundred meters to 170 km from the coastline. Large variations in the broadband noise level are observed as a function of time with the power during the noisiest time periods being more than 100 times higher than during the quiet times. Many of these noise variations are directly related to regional environmental changes (e.g., hurricane Lily and a “nor-easter”) and changes in ocean wave heights observed near Duck, NC); however, some seismic noise variation is uncorrelated with local environmental variables, indicating much more distant sources. The correlation with the ocean wave heights is better at higher frequencies (0.2–0.4 Hz) than at lower. Looking more closely, the “double-frequency” (0.15–1 Hz) and “single frequency” (0.05–0.15 Hz) microseisms show interesting patterns. At certain times, the variations of the power in the two spectral bands are independent of each other, but at other times the power increases occur across the entire 0.05–1 Hz frequency band. When broadband increases in the noise power are observed, the higher frequency noise increase always precedes the lower frequency noise increase, and the noise peak in 0.09–0.2 Hz band typically lags the peak in the 0.4–1.0 Hz band by about 7 h. In several instances, ocean wave heights display similar reddening of the spectra with time, indicating an offshore origin for the low-frequency microseismic noise and reflecting the longer time required by the regional offshore storm systems to generate low-frequency waves. The high-frequency (0.2–1 Hz) band is dominated by a number of peaks lasting about 12 h, primarily during daylight hours, and is also correlated with the local sea state. The character of these peaks indicates a regional offshore source in response to increases in wind-speed during daylight hours. Preliminary frequency–wave-number analysis indicates clear directionality for noise associated with the noise peaks that are not correlated with the local sea state. This indicates a dominant source at a large distance from which seismic waves travel essentially as plane waves across the array. The seismic noise field, however, becomes highly heterogeneous when the noise increase is correlated with the local sea state and is presumably of regional origin. In that case, the energy arrives from many azimuths along the coast line.

3:00

7A06. VLF/ULF noise on basalt and sedimented seafloor. Brian T. R. Lewis (School of Oceanography, WB-10, Univ. of Washington, Seattle, WA 98195)

During project NOBS (7 Sept. 1991 to 10 Oct. 1991) four arrays of

ocean bottom seismometers were deployed to record seafloor noise in the band 0.01 to 10 Hz. The arrays had 0-, 10-, 100-, and 1000-m sediment, and were all within a few hundred km of FLIP, which was located at 43° 42.63 N, 125° 59.19 W. Examination of two-dimensional wave-number spectra of waves from an earthquake and from noise shows that for frequencies from 0.1 to 0.5 Hz the wavelength is of order tens of kilometers, whereas above 0.5 Hz the wavelength of noise and signal-generated noise is of order tens of meters. Four conclusions can be drawn from the data at this time; from 0.1 to 0.5 Hz the noise is propagating at a relatively high velocity, probably as Rayleigh waves with speeds of order 3 km/s; from 0.5 to 10 Hz the noise is propagating at low speeds, probably as interface (Stonely) waves with speeds of order 0.1 km/s; shear energy undergoes a large amount of scattering from 0.5 to 10 Hz; noise from 0.5 and 10 Hz is probably produced by scattering at the seafloor and is largest where the sediment is thinnest.

3:15

7A07. Void-fraction measurements in bubble plumes generated by 2-D and 3-D breaking waves. Eric Lamarre (R. M. Parsons Lab., Bldg. 48-108, MIT, Cambridge, MA 02139) and W. K. Melville (Scripps Inst. of Oceanography, U.C.S.D. and R. M. Parsons Lab., MIT)

Recent experiments have confirmed that low-frequency sound (10 to 300 Hz) is generated under breaking waves. It has been suggested that volume oscillations of the bubble plume generated by breaking is the mechanism responsible for the generation of this sound. Confirmation of this process requires independent measurement of the void fraction, and therefore sound speed, in the bubbly mixture. Detailed measurements of the evolution of the void-fraction field in bubble plumes generated by 2-D and large scale 3-D laboratory breaking waves are presented and compared. Various moments of the bubble plume void-fraction field give information on the (a) volume of air entrained, (b) cross-sectional area of the plume, (c) mean void fraction and sound speed, (d) bubble plume resonant frequencies, (e) potential energy of the bubble plume, (f) horizontal and vertical displacement and velocity of the bubble plume. These measurements along with independent measurements of the sound radiated (Loewen and Melville, this meeting) support the hypothesis that low-frequency sound is generated by the volume oscillation of the bubble plume. The radial dependence of the bubble plume's void-fraction (and sound speed) field and its effect on low-frequency sound generation is investigated and the influence of the void-fraction instrumentation detection threshold in estimating the above moments is also examined. [Work supported by ONR (Ocean Acoustics) and NSF (Physical Oceanography).]

3:30

7A08. High-speed bubble sizing. D. Koller, Y. Li, P. M. Shankar, and V. L. Newhouse (Dept. of Elec. Comput. Eng., Drexel Univ., 32nd & Chestnut Sts., Philadelphia, PA 19104)

Since a breaking wave has such a dramatic effect on the bubble population over such a brief period of time, it is necessary to acquire a bubble profile in a fraction of a second. A high-speed bubble-sizing technique has been developed that should have the capability of accurately determining bubble populations during breaking waves. Testing was performed in a pool 3 m in diameter and 1 m deep. An artificial bubble cloud was created through the use of compressed air and hypodermic needles. A three-transducer system was used to determine the existence of various bubble sizes. The system developed to date was able to acquire an average bubble spectrum from 2.5–6 kHz in 0.1 s. Through the use of the double frequency technique, there would be very little or no contribution from nonresonant bubbles at each frequency of interest. The results of the system were compared to the data obtained by listening to ambient bubble noise with a transducer and from photographic population counts. These comparisons demonstrated the effectiveness of this technique. [Work supported by ONR.]

4:00

7AO9. Enhancement of hydrodynamic flow noise radiation by the regulation of air bubbles in a turbulent water jet. Murray S. Korman (Dept. of Phys., U.S. Naval Academy, Annapolis, MD 21402), Ronald A. Roy (Univ. of Washington, Seattle, WA 98105), and Lawrence A. Crum (Natl. Ctr. for Phys. Acoust., Coliseum Dr., University, MS 38677)

Experimental results show that the near-field hydrodynamic radiated flow noise (generated by a turbulent submerged circular water jet) is enhanced when the turbulent flow is modified to become a two-phase flow containing air bubbles. Acoustic intensity spectra, in the frequency band between 20 and 7000 Hz, are measured using a digital spectrum analyzer from signals generated by a hydrophone placed at the position of $Z = 4d$ and $R = 4d$. Here, Z and R are the axial and radial positions from the nozzle exit, respectively. The water velocity is 12 m/s at the nozzle exit (of diameter $d = 0.635$ cm). An amplification factor defined by the ratio of intensities $I_{\text{two-phase flow}}/I_{\text{fluid flow}}$ is measured as a function of the void fraction β of the air bubbles in an effort to verify the theoretical amplification predictions made by Crighton and Ffowcs-Williams [J. Fluid Mech. **36**, 585-603 (1969)] and more recently by Prosperetti [J. Acoust. Soc. Am. **84**, 1042-1054 (1988)]. A short videotape of the experiment will be shown. [Work supported by the National Center for Physical Acoustics, ONR, and the Naval Academy Research Council.]

4:15

7AO10. Δk -acoustic sensing of the ocean surface. A. J. Palmer, S. A. Frisch, and S. F. Clifford (NOAA Wave Propagation Lab., 325 Broadway, Boulder, CO 80303)

A new acoustic remote sensing method for measuring ocean surface directional wave spectra and currents is put forward. The method is termed Δk sonar or Δk sodar for beneath surface and above surface applications, respectively. The method is analogous to Δk radar. The basis of the Δk method is the use of two coherent beams of slightly different frequencies to project a (moving) fringe pattern onto the surface. This fringe pattern selects a single surface wave vector that will modulate the entire footprint of the beams on the surface. This modulation is revealed by the appearance of a " Δk resonance line" in the autocovariance spectrum of the frequency difference signal in the back-scattered fields. Signal-to-noise calculations are presented which indicate that the method should be practical for sensing surface currents and wave spectra out to ranges limited by attenuation of the acoustic signals.

4:30

7AO11. Simultaneous observations of acoustic scattering and ocean microstructure in the arctic. Albert J. Plueddemann (Woods Hole Oceanographic 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 Oceanography, La Jolla, CA)

Intriguing new observations from an ice camp, manned during the Cooperative Eastern Arctic Experiment (CEAREX) on the Northwest flank of the Yermak Plateau, indicate that ocean microstructure may be detectable with "standard" acoustic instrumentation (i.e., acoustic

Doppler current profilers) of moderately high frequency (150-300 kHz). The strength of the CEAREX data set is the simultaneous observation of kinetic energy dissipation rate, temperature dissipation rate, and acoustic backscatter from both 160- to 300-kHz Doppler profilers. The turbulence levels observed during CEAREX were particularly strong ($\epsilon > 10^{-7}$ W kg $^{-1}$) and occurred in well-defined patches. Back-scattered intensity anomalies of 2 to 6 dB were found to be coincident in time and space with patches of strong turbulence in the thermocline. It is shown that the acoustic intensity anomalies co-vary with the strength of the turbulent dissipation above a threshold that presumably represents 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 anomalies.

4:45

7AO12. Geomorphic parameter estimations with neural networks using bathymetry and backscatter data. Jerald W. Caruthers and Brian Bourgeois (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

A bistatic scattering strength model (BISSM) for low-frequency acoustics, which uses high-resolution geomorphology as input parameters, has been proposed [Caruthers *et al.*, NOARL, SP023:200:90 (1990)]. Included in these geomorphic parameters are (1) deterministic bathymetry and local bottom-facet mean slope and azimuth, (2) the stochastic parameters, rms slopes in orthogonal directions and roughness, and (3) the empirical acoustic parameter, Lambert/Mackenzie scattering coefficient. The issue of how to obtain these parameters to support wide-area applications of the model led to a consideration of the inverse problem using the model itself at higher frequencies and in the backscatter direction. Swath sonar systems, simultaneously providing high-resolution bathymetry and backscattering strength as a function of grazing angle, would appear tractable as survey tools for these parameters, if the inverse problem is solvable. Presented here is a sensitivity analysis using neural networks to determine the potential validity of this approach. The model is presumed to be valid and is used to simulate noise-free backscatter data for different sets of parameters. Presented is the ability of neural networks to be trained to provide estimates of the desired parameters under these ideal conditions. [Work is supported by CNOC.]

5:00

7AO13. Acoustic emissions of toroidal bubbles. Ali R. Kolaini, Michael Nicholas, and Lawrence A. Crum (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, Oxford, MS 38677)

Toroidal bubbles can be formed by injecting a small volume of air impulsively through a nozzle placed underwater. As the air leaves the nozzle, the front face (which is initially moving with a large velocity) slows down within a short distance and time, and a jet of air moving from the back of the bubble at high speed then penetrates the front face, thus forming a toroidal bubble. These bubbles then move upwards with their plane perpendicular to the direction of motion. The ring radius increases while the cross-sectional area of the air core decreases; simultaneously, the fluid velocity on the surface of the toroid slows down due to viscous effects. The combination of these two effects causes the toroidal bubbles eventually to become unstable and to break into a number of small bubbles. This phenomenon can be observed for a number of

conditions. During the formation of the toroid, low-frequency, relatively high-amplitude damped acoustic emissions were observed, with a shift to higher frequencies as a function of time. During toroid breakup, a number of bubbles within a fairly narrow size distribution were observed which emitted sound at higher frequencies and lower amplitudes than the toroid-formation sound. In this presentation some preliminary results are shown of the acoustic emissions of toroidal bubbles formed using a variety of nozzle diameters, air injection velocities and air volumes. [Work supported by ONR and ONT.]

5:15

7A014. Optical detection of transient bubble oscillations associated with the underwater noise of rain. John S. Stroud and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

It is known that for a drop of water falling on a water surface a bubble will be entrained for certain drop diameter-impact velocity combinations [Pumphrey *et al.*, *J. Acoust. Soc. Am.* **85**, 1518-1526 (1989)]. Due to the small size of the entrained bubble, the frequency of oscillation (near 14 kHz), and the very small oscillation amplitude, it is extremely difficult to view the volume pulsations of the newly created bubbles. Such oscillations contribute to the underwater noise of rain. A method was developed for optical detection of these oscillations. The entrained bubble is in the path of a laser beam when the bubble is created. The beam was then directed to a photodetector and the transient signal resulting from oscillations of the optical cross section of the bubble was recorded. This optically obtained record was compared to an acoustic record of the same event (obtained by using a hydrophone). The two records agree remarkably well in both time and frequency. The general magnitude of the initial radial oscillation is estimated and compared with theoretical results by H. N. Oguz and A. Prosperetti [*J. Fluid Mech.* **228**, 417-442 (1991)]. [Work supported by ONR.]

5:30

7A015. Broadband low-frequency sound radiator with high-frequency pump resonator. Dimitri M. Donskoy (Davidson Lab., Stevens Inst. of Technol., Castle Pt. Sta., Hoboken, NJ 07030)

A new type of underwater low-frequency radiator is described. Operation of the radiator is based on the nonlinear conversion of high-frequency acoustic pump energy to low-frequency sound pulsation of the water volume of the open acoustic pump resonator. High-intensity pump waves in the resonator produce cavitation. Water containing bub-

bles is highly nonlinear (hundreds and even thousands of times more than pure water), and dispersive. Both of these factors, and also the use of the pump resonator, promote much stronger conversion of pump energy to low-frequency sound in comparison with a parametric radiator. The calculation of the acoustical characteristics is presented.

5:45

7A016. Monterey Bay acoustic environmental monitoring system. Khosrow Lashkari (MBARI, 160 Central Ave., Pacific Grove, CA 93950), Warren W. Denner (EOS Res. Associates), James Miller, and Ching Sang Chiu (Naval Postgraduate School)

Over the past few decades sound has been used with increasing success to measure and monitor a variety of oceanographic features and processes—biological populations, water structure, internal waves, bathymetry, and climatic change to name just a few. Monterey Bay and surrounding ocean regions have recently been proposed as a Marine Sanctuary. Scientists see this region as a natural marine laboratory. In an effort to unobtrusively monitor features and processes in this ocean region a group of local scientists have initiated the design of an Acoustic Environmental Monitoring System (AEMS). A preliminary design meeting was held 28-29 February 1992 by a group of interested scientists and engineers at the Monterey Bay Aquarium. This paper reports on the preliminary design and objectives of the AEMS based on that meeting. The AEMS will consist of three monitoring sites within Monterey Bay, one near the head of Monterey Canyon, and two on the adjacent shelf areas. The sites will be connected to shore by high bandwidth telemetry links. Each site will consist of an instrumentation platform, with quick connect sensor mountings. While acoustic instrumentation will constitute the backbone, the monitoring systems will have provisions to support other oceanographic instruments. Hardware and software interface standards will be established for accommodating these sensors. The acoustic instrumentation will consist of hydrophone arrays, projectors, and acoustic Doppler current meters. The intent is to provide a long-term monitoring capability and a test bed for a variety of environmental monitoring systems. Some of the initial objectives for the AEMS are: provide a long-term acoustic research facility available to the civilian scientific community; provide a reception site for global-warming experiments and other long-range ocean propagation research; monitoring activity in the Marine Sanctuary; marine mammal research; monitoring geological processes; ocean circulation and water column processes. The purpose in presenting this paper at the May 1992 meeting of the Acoustical Society of America is to gain input to this ambitious project from the largest gathering of knowledgeable scientists and engineers possible.

Session 7BV

Bioresponse to Vibration and Physical Acoustics: Ultrasonically Induced Cavitation in Biological Systems

Floyd Dunn, Chair

Bioacoustics Research Laboratory, University of Illinois, 1406 West Green Street, Urbana, Illinois 61801

Chair's Introduction—1:00

Invited Papers

1:05

7BV1. Acoustic cavitation from diagnostic ultrasound. Robert E. Apfel, Christy K. Holland (Ctr. for Ultrason. and Sonics, Yale Univ., New Haven, CT 06520), Lawrence A. Crum (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, Oxford, MS 38677), and Ronald A. Roy (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Over the past several years a collaborative effort has been undertaken by several research groups to investigate the probability of *in-vivo* acoustic cavitation by diagnostic ultrasound devices. In the course of this research, movement has progressed from the initial prediction of its existence, through a variety of experiments *in vitro* using existing clinical devices in which the characteristics of this cavitation have been measured, to the current efforts in which *in-vivo* experiments have been attempted. There will be a joint review of past progress and future plans will be described. [Work supported by NIH through Grant No. CA 39374.]

1:35

7BV2. Apparent contribution of respiratory gas exchange to the *in-vitro* "cell density effect" in ultrasonic cell lysis. Andrew A. Brayman (Dept. Biophys., School of Med. and Dentistry, Univ. of Rochester, Rochester, NY 14642), Yukio Doida (Shiga Univ., Japan), and Morton W. Miller (Univ. of Rochester)

Ultrasonic cell lysis *in vitro* varies dramatically with cell volume density; densely suspended particles are thought to interfere with gas body activity. The general postulate that the "cell density effect" can be explained in part by cellular modification of the suspension medium by respiratory gas exchange was experimentally tested. Ultrasonic cell lysis occurred at higher levels in suspensions supplemented with the respiratory inhibitor NaCN than in control suspensions when suspension densities exceeded 2×10^7 cells/ml. At constant cell density, cell lysis diminished with increasing pre-insonation incubation time at 37 °C; the rate of this change was diminished significantly by cyanide treatment. These observations are consistent with the postulate that respiratory O₂:CO₂ exchange enriches the medium in CO₂ while depleting the medium of O₂, which cavitates more readily than CO₂, thereby diminishing the potential for cavitation-related cell damage. However, this is only a partial explanation of the cell concentration dependence of cell lysis; cell density *per se* is an important factor in the phenomenon.

2:00

7BV3. A comparison of hemolytic and sonochemical activity of ultrasonic cavitation in a rotating tube. Douglas L. Miller and Ronald M. Thomas (P7-53, Battelle PNL, P.O. Box 999, Richland, WA 99352)

Cavitation bioeffects of ultrasound *in vitro* may potentially result either from mechanical or from sonochemical mechanisms. Mechanical hemolysis was assessed in a 60-rpm rotating-tube system with unfocused 1.61-MHz exposure by the spectrophotometric method. Free radical generation was assessed by the terephthalic acid dosimeter, calibrated by gamma-ray dosage. Sonochemical production was assessed by measuring residual hydrogen peroxide using the isoluminol/microperoxidase method. Longer durations were needed for the free radical (e.g., 4 min) and hydrogen peroxide (e.g., 15 min) tests than for the hemolysis tests (e.g., 16 s). The sonochemical mechanism was relatively more important for increasing intensity (2.8 to 11 W/cm² SpTa), and increasing temperature (2, 12, 20, and 37 °C). Bubbling with argon before exposure appeared to enhance both mechanisms. Burst mode exposure (10-μs bursts, 1:1 or 1:3 duty cycles) reduced cavitation activity, but gave relatively greater sonochemical activity for constant temporal average intensity. The results should be helpful for selecting exposure conditions suitable for studying

2:25

7BV4. Cavitation bioeffects: Spin-offs from lithotripsy. Edwin L. Carstensen (Ctr. for Biomed. Ultrasound and Dept. of Elec. Eng., Univ. of Rochester, Rochester, NY 14627)

Lithotripter fields can produce cavitation in the organs of the body at pressure amplitudes significantly lower than those required to fragment stones. Because of limits on the expansion of the gaseous nuclei imposed by the surrounding tissue structures, cavitation in tissue is qualitatively different from that described by classical cavitation theory for a spherical bubble in an infinite fluid. Thresholds for structural and functional changes in kidney, liver, lung, heart, and blood, developmental effects in the embryo/fetus, and killing of *Drosophila* larvae have been determined. Studies of the biological effects of shock waves generated by lithotripters have guided us in the search for effects of diagnostically relevant pulsed ultrasound. In particular, lung hemorrhage occurs with pulsed ultrasound at acoustic pressures of the order of 1 MPa and effects on the dynamic behavior of heart tissue have been seen with single 1–10 ms pulses at pressure amplitudes of the order of 10 MPa. The effects in lung occur at temporal average intensities orders of magnitude lower than previously associated with biological effects in mammals.

2:50–3:05

Break

3:05

7BV5. An examination of echocontrast enhancement by frequency-modulated acoustic pulses. J. B. Fowlkes, J. X. Tan, P. L. Carson, and J. M. Rubin (Dept. of Radiology, Div. of Radiological Phys. and Eng., Univ. of Michigan Med. Ctr., Ann Arbor, MI 48109-0553)

Albumin-encapsulated microbubbles (3- μ m median diam) suspended in a 5% bovine albumin solution, or in canine urine or whole blood were subjected to 1-s acoustic pulses with pressure amplitudes of up to 4 bars and frequency sweeps between 1 and 10 MHz. Experimental measurements indicated a differential increase in the frequency-dependent backscatter and attenuation when the frequency within the pulse decreased with time (referred to as a "prich") rather than increased ("chirp"). The decreasing frequency of the prich pulse would more closely follow the resonance frequency of bubbles being grown by rectified diffusion and is thus a possible explanation for the dependence of scattering and attenuation on the frequency sweep direction. Rectified diffusion theory suggests an optimum frequency modulation for maximum bubble growth. A theoretical investigation is presented which examines the effects of the bubble size distribution, rectified diffusion, radiation pressure, and other parameters on the acoustic backscatter and attenuation measured in the present experimental system and the potential efficacy of an echocontrast enhancement system in medical imaging applications. [Work supported by NIH 1R01-DK42290.]

3:30

7BV6. Generation of cavitation by extracorporeal shock waves *in vivo*. M. Delius and S. Gambihler (Inst. for Surgical Res., Univ. of Munich, Klinikum Grosshadern, Munich, Germany)

Extracorporeal shock waves are used in medicine to destruct kidney and gallstones. As a side effect, they can cause hemorrhage in tissues. During shock wave application, a transiently increased tissue echogenicity is observed by diagnostic ultrasound suggesting temporary generation of gas. When shock waves were applied to piglet livers, gas bubbles persisted for several hundred milliseconds in liver vessels. Three different findings suggest that the interaction between gas bubbles and shock waves is involved in the generation of tissue damage: First, the extent of renal hemorrhages depended on the rate of shock wave administration. Damage was most severe when the identical number of shocks was administered at a rate at which gas bubbles had not been flushed out of the high pressure field by the blood flow. Second, Prat *et al.* have demonstrated that injection of air microbubbles into the hepatic artery of rabbits greatly enhanced tissue damage. Third, tissue damage occurred in canine livers at the same sites as focal and transient accumulations of gas bubbles.

7BV7. Exposure levels for lung hemorrhage and other bubble effects *in vivo*. Leon A. Frizzell (Dept. of Elec. and Comput. Eng., Univ. of Illinois, 1406 W. Green St., Urbana, IL 61801)

The lungs of neonatal mice were exposed *in situ* to 1-MHz pulsed ultrasound (10- μ s pulse duration, 1-kHz pulse repetition frequency) for durations of 2.4 and 180 s at 10 °C. The neonates were mounted in a special holder that was placed in a tank of physiologically compatible coupling medium such that the broad focal region of the transducer was centered on the dorsal surface overlying the lungs. After removal of the animal from the holder the lungs of sonicated and sham control animals were carefully removed and examined under a dissecting microscope by a second person, who did not know whether the specimen was a sham or sonicated animal, for signs of hemorrhage. The threshold levels for lung hemorrhage are well below those for bubble effects reported in other tissues, but in good agreement with levels reported for hemorrhage in adult mouse lung [S. Z. Child *et al.*, *Ultrasound Med. Biol.* **16**, 817-825 (1990)]. The dependence upon on-time for the lung hemorrhage was similar to that for bubble effects in other tissues.

Contributed Papers

4:20

7BV8. Measurement of cavitation thresholds for human blood. Qihong Xu, Christy K. Holland, and Robert E. Apfel (Ctr. for Sonics and Ultrason., P.O. Box 2159, Yale Univ., New Haven, CT 06520)

To gauge the likelihood of producing transient microcavitation *in vivo* with medical ultrasonic equipment, several cavitation experiments were conducted with bank blood using acoustic signals that are similar to those deployed by typical medical diagnostic ultrasound. Using a specially designed chamber, filled with bank blood samples having different volume concentrations of red blood cells (RBC), the cavitation threshold of the blood and the backscattering signal from the blood were measured with an active cavitation detector [Roy *et al.*, *J. Acoust. Soc. Am.* **87**, 2451-2458 (1990)] with the frequency centered at 30 MHz. The results show that the acoustic pressure causing microcavitation in blood is in the range of the pressure generated by some medical diagnostic devices, the cavitation threshold of the blood varies linearly with the volume concentration of the RBC, and the noise level due to backscatters is inversely proportional to the volume concentration of the RBC. [Work supported by NIH Grant No. 5R01CA39374.]

4:35

7BV9. A comparison of mechanical lysis of erythrocytes and Chinese hamster ovary cells induced by ultrasonic cavitation in a rotating tube. Ronald M. Thomas and Douglas L. Miller (P7-53, Battelle PNL, P.O. Box 999, Richland, WA 99352)

Hemolysis of erythrocytes was compared to the lysis of cultured mammalian cells to determine how well erythrocytes can substitute for nucleated cells for the investigation of mechanical cell damage induced by ultrasound. Canine erythrocytes and Chinese Hamster Ovary (CHO) cells were exposed to cavitation in a 60-rpm rotating tube with a 1.61-MHz unfocused transducer, at increasing intensities (2, 4, and 8 W/cm² SPTA), both separately at 2.5×10^6 cells/ml and mixed together, each with 1.25×10^6 cells/ml. Cell viability was assayed by counting on a hemacytometer, intact cells which excluded trypan blue dye. Results were normalized to 100% viability in sham exposed samples. The CHO cells appeared to be somewhat more sensitive to lysis under some conditions, and significant differences were noted after six repetitions at 8 W/cm² in the separate and in the mixed exposures. At this intensity, viability was 73% (separate) and 58% (mixed) for the erythrocytes and 37% (separate) and 32% (mixed) for the CHO cells. [Supported by PHS Grant No. CA42947 awarded by the National Institute of Health.]

4:50-5:30

Bull Session

Session 7MU

Musical Acoustics: The Singing Voice

Ingo R. Titze, Chair
330 SCH, University of Iowa, Iowa City, Iowa 52242

Chair's Introduction—1:30

Invited Papers

1:35

7MU1. Acoustical profile of trained low male classical singing voices ranked as to perceptions of relative "beauty." Clayne W. Robison, Barry Bounous, and Ross Bailey (Dept. of Music, Brigham Young Univ., E-461 HFAC, Provo, UT 84602)

Based upon a uniform 20-s musical excerpt, a score of classically trained lower male singers will be ranked as to vocal "beauty" by a series of "expert" witness groups: (1) a panel of nationally published classical voice instructors, (2) a panel of classical voice instructor colleagues from a single unified voice faculty (with this group complete anonymity could not be preserved), and (3) a panel of knowledgeable, but amateur opera listeners. Correlation studies will be done among the witnesses in each "expert" group and between groups. From those correlations acoustical spectrum comparisons will be made among the ranked singers to determine which acoustical parameters seem most consistently present in those singers ranked as most "beautiful." From a simultaneous video-taping of the sung excerpts, hypotheses concerning the physiological (postural) and aerodynamic events which led to those acoustical parameters will be spun out. This lecture will report the results of this study and discuss the related hypotheses.

2:05

7MU2. Respiratory control during singing. Thomas J. Hixon and Peter J. Watson (Natl. Ctr. for Neurogenic Commun. Disord., Univ. of Arizona, Tucson, AZ 85721)

Singing presents special control tasks for the respiratory apparatus. These tasks are performed variously by singers and have been studied through the use of several technologies. This paper will discuss data gathered over the past decade in this laboratory on the nature of respiratory function during singing and interpret those data in terms of contemporary theory about neuromotor and biomechanical features of respiratory control. As well, attention will be devoted to respiratory control issues surrounding the development of singing skill and the potential to reorganize singing performance under selected biomechanical constraints. [Work supported by the National Institute on Deafness and Other Communication Disorders.]

2:35

7MU3. Acoustics of the tenor high voice. Ingo R. Titze (Dept. of Speech Pathology and Audiology and Natl. Ctr. for Voice and Speech, Univ. of Iowa, Iowa City, IA 52242 and The Recording and Res. Ctr., The Denver Ctr. for the Performing Arts, Denver, CO 80204) and Sharon Mapes (Westminster Choir College, Princeton, NJ)

Frequency spectra of five standard vowels sung at high pitches (F_4 , G_4 , A_4 , and B_4) were obtained from six tenors who were in vocal training at Westminster Choir College. Because of the variation of the spectra over the vibrato cycle, measurements were made at the peak, the trough, and the middle of the cycle. Stylized power spectra were generated from averages over the vibrato cycle. A power spectrum model was then used to determine the extent to which tenors "tune" formants to specific harmonics of the source. Results indicate that, aside from the high level of acoustic energy in the region of the singer's formant, tenors get most of their vocal intensity from the second, third, and fourth harmonics. In some cases, F_1 or F_2 is specifically tuned to one of these lower harmonics; in other cases, however, the desire to create smooth transitions between vowels seems to favor a distribution of energy across several harmonics. Unlike what has been observed in the soprano high voice, the fundamental almost never carries the greatest amount of energy.

7MU4. Aperiodicities in the singer voice source. Perry R. Cook (Stanford Ctr. for Comput. Res. in Music and Acoust., CCRMA, The Knoll, Stanford Univ., Stanford, CA 94305)

Aperiodicities in the glottal source waveforms of singers were investigated. Fast, interactive signal processing tools were developed for investigating aperiodicity. The expected deviations at the instant of glottal closure were found, along with two other commonly occurring types of disturbance. Pulses of noisy disturbance near the moment of glottal opening were often observable in low male tones. It was hypothesized that this noise production is caused by local turbulence near the glottal source which is modulated by the source flow wave itself. Analysis results for several bass singers producing multiple vowels across the vocal range were consistent with the dependence of the turbulence modulation on frequency and tube diameter downstream from the source. Subharmonic generation, often called diplophonia and considered a vocal pathology, is quite common and natural in normal singer voices. Subharmonic generation was investigated in 20 singers, across four octaves, three vowels, and three dynamic levels. Each subject exhibited the presence of measurable subharmonics on at least one produced tone, with over one third of all produced tones containing a second subharmonic component.

3:35-3:45

Break

Contributed Papers

3:45

7MU5. Making and hearing the vocal trill: An acoustical study. Michèle Castellengo and Damien Colas (Lab. d'Acoust. Musicale, Univ. Paris 6 et CNRS URA 868, Tour 66, 4 Place Jussieu, 75005 Paris, France)

Among vocal ornaments trill is the least well known. Several records of trills done by great singers have been analyzed in order to define and measure relevant characteristics for this ornament: the amplitude of the vibrato during the preparation, the extension and speed of the trill itself, the termination pattern. The importance of the accompanying amplitude modulation has also been investigated. An hypothesis for the particular vocal technique of bass singers is proposed. These various acoustical parameters are used as data for digital synthesis based on formant-waveform algorithm. Artificial vocal sequences are tested by subjects in order to evaluate the perceptual effect of the variations of the acoustical parameters and to appreciate the relative importance of the different elements of a trill. These experiments have pointed out the great importance of the preparation phase for perceiving a trill, and appreciating its interval (major or minor).

4:00

7MU6. A parametric study on pitch of short vocal vibrato tones. Christophe d'Alessandro (LIMSI-CNRS, BP 133-91403 Orsay-Cedex, France) and Michèle Castellengo (LAM, Univ. of Paris, VI, 4 Place Jussieu 75005 Paris, France)

The pitch perceived for short vocal vibrato tones was measured using a method of adjustment. The stimuli were synthetic vocal tones, produced by a formant synthesizer. The main parameter under study was the tone duration, as a function of the fractional number of vibrato cycles. This parameter was examined in relation to: (1) the vibrato extent (0, 50, 100, and 200 cents); (2) the vibrato rate (4, 6, and 8 Hz);

(3) the tone frequency (220, 440, 880, and 1500 Hz). Durations from $\frac{1}{2}$ cycle to 5 cycles were studied. Part of the results were obtained for a relatively large group of musically educated subjects (20 subjects), and another part for a small group of selected subjects. Our results show that: (a) for short tones, the pitch does correspond to a weighted time average of the $F0$ pattern (a numerical model of which is in accordance with this data); (b) the pitch mean between the extreme frequencies as the duration increases; (c) the overall pattern of $F0$ has an influence on perception, and some simple patterns seem to behave better perceptually; (d) perception may be ambiguous above a threshold of duration, which is related to the absolute threshold of pitch change and to the trill threshold.

4:15

7MU7. Advanced methods for voice analysis. Yingyong Qi (Dept. of Speech and Hear. Sci. and Dept. of Elec. and Comput. Eng., 106 Speech Bldg., Univ. of Arizona, Tucson, AZ 85721)

The harmonics-to-noise ratio (HNR) has been widely accepted as a measure for quantifying the irregular or noise component of voice. The use of the HNR assumes that: (1) the noise or irregularity in voice is a stationary process, and (2) errors introduced by the cycle-to-cycle pitch period variations are trivial. When either of these assumptions is violated, the HNR is inappropriate for estimating irregularities in the shape of acoustic wavelets. Methods for eliminating these assumptions (or limitations) of the original HNR have been developed based on advanced techniques of signal processing. In these methods, nonstationary irregularities in voice are identified using an adaptive Wiener filter. The effects of pitch period perturbations on the measurement of wavelet irregularities are minimized by optimal time normalization of wavelet assemble using procedures of dynamic time warping. These methods have been evaluated using natural and synthetic voices and results have indicated that they significantly increase the accuracy and reliability of voice analysis.

Panel Discussion

This panel discussion will highlight some of the acoustic and physiologic principles involved in traditional and nontraditional styles of singing. The discussion will follow a live demonstration and will include the audience.

PANEL MODERATOR: Ingo R. Titze

PANEL MEMBERS: Perry Cook
Thomas Hixon
Clayne Robison

THURSDAY AFTERNOON, 14 MAY 1992

SALON F, 1:10 TO 5:00 P.M.

Session 7PP

Psychological and Physiological Acoustics: Discrimination of Complex Sounds and Speech Perception

Richard R. Fay, Chair

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

Chair's Introduction—1:10

Contributed Papers

1:15

7PP1. Detection of scattered ambient noise by fish. Thomas N. Lewis and Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

It has been hypothesized that one role of the fish's auditory system may be to detect and localize nearby fish by "imaging" ambient noise scattered by their swimbladders. This is analogous to the role of the visual system of most animals, where the relevant signal is ambient light scattered by objects rather than light emitted by luminous objects. A classical conditioning experiment has been performed which indicates that the fish auditory system is capable of functioning in this manner. The ambient noise was provided by a J-9 transducer driven by Gaussian noise. Scattering of ambient noise by the resonant swimbladder was simulated by applying a filtered version of the same Gaussian noise to a small spherical projector. The target strength and bandwidth of the scattered noise were controlled by the attenuation and passband of the filter. The fish was conditioned to respond to the presence of the signal from the spherical projector. Thresholds of simulated scattered noise as a function of range to the subject were measured for the species *Carassius auratus* (common goldfish). The fish's ability to discriminate this signal is taken to be a measure of its ability to detect scattered ambient noise. [Work supported by ONR.]

1:30

7PP2. Analytic listening in the goldfish: Simultaneous frequency analysis. R. Fay (Parmly Hear. Inst. and Dept. of Psychol., Loyola Univ. of Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

A stimulus generalization paradigm was used with classical respiratory conditioning to determine the extent to which two simultaneous tones are analyzed separately, or "heard out" by the goldfish without explicit differential conditioning. Four groups of animals were used. Two groups were conditioned to suppress respiration to a 6-s pure tone (166 or 724 Hz). They were then tested for generalization to novel tone frequencies from one octave below to one octave above the conditioning frequency. The magnitude of respiratory suppression peaked at the conditioning frequency and declined monotonically toward higher and lower frequencies. Two other groups were conditioned to either 166- and 724-Hz tones alternating on successive trials, or 166 and 724 Hz mixed on every trial at equal sensation levels as a two-tone complex. Both groups were tested identically for generalization to novel pure tones from 79 to 1500 Hz. The generalization functions of frequency for both groups were nearly identical: Suppression responses were largest at the conditioning frequencies and progressively declined at more remote frequencies, producing generalization functions with peaks at the tone frequencies used in conditioning. These results demonstrate that the goldfish acquires independent information about the frequency components making up a complex tone, and thus listens analytically. The

resolution of simultaneous frequency analysis is no less than for the same pure tones presented separately. This is the first demonstration of analytic listening in a nonhuman vertebrate. [Supported by a Center Grant from the NIDCD.]

1:45

7PP3. Applying the sonar equation to the detection of a bottom target by an echo-locating dolphin. Whitlow W. L. Au (Naval Ocean Systems Ctr., P.O. Box 997, Kailua, HI 96734)

The target detection range of two echolocating Atlantic Bottlenose dolphins as a function of target depth in Kaneohe Bay, Hawaii was determined by Murchison [A. E. Murchison, Ph.D. dissertation, Univ. of Calif. Santa Cruz (1980)]. The threshold range decreased monotonically as the depth of the 6.35-cm-diam solid-steel sphere increased and approached the bottom. For the target lying on the bottom, the threshold detection range was approximately 70 m. The scattering strength of the bottom in Kaneohe Bay at approximately the same location of the Murchison's experiment was recently measured using a simulated dolphin echolocation signal and a transducer tilted at the appropriate grazing angle. The bottom scattering strength along with the target strength of the 6.35-cm sphere and the dolphin threshold range were incorporated into the generalized form of the sonar equation for a reverberation-limited situation and a detection threshold of 4.0 dB was calculated. This detection threshold compared well with the 2.3 dB obtained in an experiment in which the dolphin was required to detect a target in the presence of a clutter screen [W. Au and C. Turl, *J. Acoust. Soc. Am.* **73**, 1676-1681 (1983)].

2:00

7PP4. "Within" and "between" sound comparisons. Irwin Pollack (Mental Health Res. Inst., Univ. of Michigan, Ann Arbor, MI 48109-0720)

An isolated sound can be judged to be of "high pitch or low pitch" without direct explicit comparison with another sound. However, the comparative judgment of "higher than or lower than" implies a comparison *between* two sounds, although the two sounds need not be physically presented together, as in a one-interval test. On the other hand, the judgment of whether an isolated sound is harmonically "pure or impure" can be based upon the information *within* an isolated sound, and, in theory at least, does not imply a comparison between sounds. The discrimination of frequency differences within the mixture of two complex periodic signals (repeated random waveforms) was examined in a one-interval test. When the fundamental frequencies of the components within the mixture have a small-numbered ratio, the harmonic "purity" of the mixture could be judged. Judgments of harmonic purity were substantially more accurate than judgments of equivalent frequency differences, presumably in terms of pitch.

2:15

7PP5. Attending to multiple discrimination cues. Theodore Venema and B. Espinoza-Varas (Dept. of Commun. Disord., Univ. of Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

Measures of listeners' ability to attend to each of four possible discrimination cues were obtained. The stimuli comprised pairs (T_1, T_2) of 1500-Hz, 80-ms, 66-dB SL tones separated by a 80-ms silent interval [Espinoza-Varas, *J. Acoust. Soc. Am.* **74**, 1687-1694 (1983)]. In a three-interval, 2AFC task, listeners discriminated a "standard" pair (interval 1) from a "comparison" pair (interval 2 or 3) containing increments in the duration (ΔT) or in the frequency (ΔF) of either T_1 or T_2 . Each of the four possible increments was controlled by an adaptive track targeting 71-percent-correct thresholds (Levitt, 1971). Separate adaptive tracks were used for each increment, but all four tracks were interleaved randomly within a block of trials. That is, only one

kind of increment occurred *on a given trial*, but all four increments alternated randomly within a *block* of trials. Each adaptive track terminated after 10 reversals, and thresholds were defined as the average increment value of the last five reversals. A block of trials terminated when the ten-reversals criterion was reached with all four cues. The slope of the adaptive tracks and the thresholds reached after ten reversals provided measures of attention distribution. [Work supported by OCAST Grant No. HSO-005 and Presbyterian Health Foundation.]

2:30

7PP6. Spectral shape discrimination with random variations in background dimensions. Xiaofeng Li and Richard E. Pastore (Dept. of Psychol., SUNY, Binghamton, NY 13902-6000)

The current research investigated the ability of observers to discriminate a global property of a complex stimulus—the slope of the linear spectral envelope as a function of variation in two background dimensions: fundamental frequency (thus pitch) and a ripple filter function imposed upon the spectral envelope. Two dimensions are judged to be integral if a significant decrement in slope discrimination performance occurs due to a roved relative to fixed background dimension. All experiments used a XAB task with roving overall spectral level within trials to eliminate intensity cues. A significant decrement due to roving fundamental frequency suggests that the linear spectral envelope is integral with fundamental frequency. In contrast, a strikingly small decrement due to roving the ripple filter function suggests that the linear spectral envelope is more easily separable from the ripple filter function than from fundamental frequency. Therefore, the global property of slope of the linear spectral envelope may be unitized with pitch, but is relatively independent of a modest (4-dB) variation in the actual spectral envelope. [Work supported by NSF.]

2:45

7PP7. The influence of spectral composition of complex tones and of musical experience on the perceptibility of virtual pitch. Annemarie Preisler (Dept. of Musicology, Karl-Franzens-Univ., Mozartgasse 3, 8010 Graz, Austria)

Fifty-eight trained musicians and 58 naive listeners were instructed to adjust pure tones, generated by a computer-soundsampler system (ATARI MEGA/STE, AKAI S1100), to the perceived fundamental frequencies of alternatively presented harmonic complexes. Stimuli varied according to the number of their partials, the relative distance to the missing fundamental, and spectral density. Every person had to fulfill 112 different items. The degree of task difficulty was determined by the average of deviations of subjects' performances from the greatest common submultiples of the complexes. It is shown that each of the considered variables has significant influence on fundamental frequency estimation. Moreover musicians tend to tune fundamentals to low (average: -17 Cent) whereas for nonmusicians the opposite trend is observed (average: +15 Cent). Results were related to performances on the subtests "Virtual/Spectral Test" [M. E. Albrecht, doctoral thesis (1972)] and "Interval-Classification" (Preisler, 1991) by a statistical ANOVA design. Interactions between performance scores are discussed with respect to their theoretical background. [Work supported by the Austrian Fonds for Scientific Research.]

3:00

7PP8. Observation of a medication-induced change in pitch perception. Vladimir Chaloupka (Phys. Dept., Univ. of Washington, Seattle, WA 98195), Richard Muirhead, and Stephen Mitchell (Hall Health Clinic, Univ. of Washington, Seattle, WA 98195)

This paper reports on a study of an absolute pitch possessor who, upon administration of the psychoactive drug Tegretol (carbamazepine), experienced a significant change in her pitch perception. The

subject's performance both in producing as well as in identifying random tones was measured, covering the period of administration of the drug, as well as control periods before and after. The main effect was a downward shift of the pitch perceived when a synthesized tone was heard, as compared to the perception during the two control periods. The magnitude of the shift was observed to increase with increasing frequency of the stimulus; the average shift was about one semitone.

Detailed results on the frequency dependence and time dependence of the pitch shifts are presented. An upward pitch shift of comparable magnitude was observed when the subject was requested to produce a musical note identified by its name. To our knowledge, this is the first documented report of a significant, reversible change of pitch perception caused by a medication. [Work partially supported by the University of Washington Graduate School Research Fund.]

3:15-3:30

Break

3:30

7PP9. Infant psychometric functions: Asymptotic performance. Jill Y. Bargones, G. Cameron Marean, and Lynne A. Werner (Dept. of Speech and Hear. Sciences, Univ. of Washington, JG-15, Seattle, WA 98195)

Psychometric functions have been obtained for detection of repeated, short (16-ms) tone bursts [L. A. Werner and G. C. Marean, *Assoc. Res. Otolaryngol.* (1991)] and for single, long (300-ms) tone bursts [J. Y. Bargones and L. A. Werner, *Assoc. Res. Otolaryngol.* (1992)] from infants and adults. Infant functions for these stimuli are similar: they tend to have shallower slopes and lower upper asymptotes than those of adults. Best infant performance for both stimuli is about 85% correct. Infant thresholds are also worse than those of adults, but this may result partially from the fact that thresholds were estimated using methods that assume an upper asymptote of 100%. However, most studies of infant threshold present a series of long duration, repeated stimuli. In the current study, upper asymptotic performance for this more optimal stimulus was examined. The observer-based psychoacoustic procedure [Olsho *et al.*, *Devel. Psychol.* 23, 627-640 (1987)] was used to test 6- to 9-month-old infants. The stimulus consisted of 1-kHz tone bursts, with 500-ms duration. The tone was presented about 25 dB above previously estimated threshold for a similar stimulus. Asymptotic performance was predicted for this stimulus. On signal trials, the tone burst was repeated four times, with 500 ms between bursts. No-signal trials consisted of 4 s of silence. A 30-trial block, consisting of 15 signal trials and 15 no-signal trials, was obtained from each infant to estimate $p(C)_{max}$. These data were used to estimate the effect of reduced upper asymptote on infant threshold estimates. [Work supported by NIH Grant No. DC00396.]

3:45

7PP10. Discrimination of frequency transitions as a function of spectral and temporal psychoacoustic cues. J. F. MacNeil and E. B. Slawinski (Dept. of Psychol., Univ. of Calgary, 2500 University Dr., Calgary, Alberta T2N 1N4, Canada)

The present study examined discrimination of short duration frequency glides (60 ms) analogous to the second formant transition of speech. An interdependence of rate of change, frequency excursion, and duration complicates the study of frequency transitions, and to determine how these psychoacoustic cues impact upon perceptual decisions stimuli in the current studies varied as a function of offset frequency and rate of change. Three continua of computer synthesized signals were used to determine just noticeable differences (jnds) for signals described by: a constant onset frequency with diverging offset frequencies; varying onset frequencies with a constant offset frequency; and varying offset frequencies with a constant rate of change. To attenuate frequency onset information, a 20 ms of octave-band noise was appended to the beginning of each signal. Results showed that discrimination improved as a

function of the available psychoacoustic information: covarying transition rate and offset frequency yielded smaller jnds than varying either offset frequency or rate of change alone. Discrimination of transitions is not a simple function of frequency excursion or rate of change and integration of rate-related and spectral properties is speculatively important in perceptual decisions regarding complex signals such as speech.

4:00

7PP11. Duration discrimination of affricate silent intervals: Performance in fixed- and roving-standard tasks. Timothy A. Boyle and Blas Espinoza-Varas (Univ. of Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

Duration discrimination for the silence preceding the frication noise in the word /watʃ/ was measured. The utterance was produced by a female talker and digitized at 20 kHz. The 200-ms affricate silence was edited to obtain: (a) seven tokens with silence ranging from 15-45 ms in steps of 5 ms ("short" range); and (b) seven tokens with silence ranging from 100-200 ms in steps of about 17 ms ("long" range). The "short" and "long" range were perceived as /waf/ and /watʃ/, respectively. With each range, discrimination was measured in both fixed-standard and roving-standard 2IFC tasks, using a constant-stimuli method. Listeners discriminated 5%-60% decrements in the duration of the silence. Thresholds were interpolated at the 75%-correct level from probits fitted to five-point psychometric functions. For the "long" silence tokens, thresholds (expressed as Weber fractions) ranged from 0.08-0.18 and 0.07-0.13 for roving- and fixed-standard tasks, respectively. For "short" silence tokens, the corresponding Weber fractions ranged from 0.09-0.82 and 0.08-0.73. [Work supported by OCAST Grant No. HSO-005 and Presbyterian Health Foundation.]

4:15

7PP12. Level discrimination of stop-consonant noise bursts by normal-hearing and hearing-impaired listeners. Cynthia D. Brown and Blas Espinoza-Varas (Commun. Disord., Univ. of Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

Level discrimination for the consonant noise bursts of the word "pack" was measured in normal-hearing and hearing-impaired listeners. The utterance, produced by a female talker, was digitized at 20 kHz, edited and output via a D/A converter. Using a constant-stimuli 2IFC procedure, listeners had to discriminate level increments ranging from 0.5-7.0 dB in noise bursts either "in isolation" (i.e., excerpted from the word) or in word context. The sensorineural impaired ears had fairly uniform losses ranging from 30-70 dB. In all conditions, burst presentation level was 59-60 dB SPL for normal ears, and 25-30 dB above the PTA for impaired ears (i.e., most of the burst's spectra were clearly above the audibility thresholds). Discrimination thresholds [$P(c) = 75$]

were interpolated from probits fitted to nine-point psychometric functions. In normal ears, thresholds were 1.5–2.0 dB for both the /p/ and /k/ bursts presented “in isolation.” Thresholds “in context” were ~2.5 dB for /k/ and 3.0–4.0 dB for /p/. Some impaired ears showed discrimination thresholds similar to those of normal ears. [Work supported by OCAST Grant No. HSO-005 and Presbyterian Health Foundation.]

4:30

7PP13. Speech feature perception in prelingual deaf children with cochlear implants. Richard S. Tyler, Holly Fryauf-Bertschy, Danielle Kelsay, and Bruce Gantz (Dept. of Otolaryngology, Univ. of Iowa, Iowa City, IA 52242)

The speech-perception abilities of 15 prelingual deaf children, using the Nucleus 21-channel cochlear implant, were examined with a new audiovisual speech-feature test for young children. The test stimuli include the letters “b, d, c, p, t, v, z” and pictures of “me,” “knee,” and “key.” Preliminary test retest data suggest good reliability for a 60-item test. Performance is significantly correlated with a closed-set picture test for young children. A feature analysis indicated that the voicing and envelope features were understood moderately well when presented by audition, and the place feature was understood well when presented by vision. [Work supported by NIH.]

4:45

7PP14. Speech and hearing applications of LabVIEW®. B. Espinoza-Varas, Chongjoon Shim, and Theodore H. Venema (Dept. Commun. Disord., Univ. Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190)

A system for research, teaching, and clinical applications in speech and hearing is described. The system utilizes LabVIEW® software, National Instruments® expansion boards (D/A converter, DIO and DMA boards), and a Macintosh® IIFX computer. LabVIEW is an integrated package for data acquisition, digital signal processing, instrument and process control, data analysis and graphics. The package: (a) employs an icon-based language that does not require writing of code; (b) contains a comprehensive library of modular, pre-programmed functions (e.g., A/D conversion, digital filtering, FFT analysis, etc.) callable by displaying icons to the screen; (c) links modules with a graphical “wiring” utility; and (d) is intuitively simple to use. The system is being used for signal generation, control of psychoacoustic and speech-perception experiments, data analysis, and signal processing. A package of hearing-science laboratory demonstrations has been implemented also. [Work supported by OCAST Grant No. HSO-005 and Presbyterian Health Foundation.]

THURSDAY AFTERNOON, 14 MAY 1992

SALON J, 1:30 TO 5:15 P.M.

Session 7SA

Structural Acoustics and Vibration: Contributed Session II

Scott D. Sommerfeldt, Cochair

Applied Research Laboratory and Graduate Program in Acoustics, Pennsylvania State University, State College, Pennsylvania 16804

Aynur Unal, Cochair

Vibration and Sound Research Institute, 1625 Alameda, San Jose, California 95126

Contributed Papers

1:30

7SA1. Axial surface waves on fluid-loaded cylindrical shells below the ring frequency. Steven L. Means and Allan D. Pierce (Graduate Program in Acoust., Penn State Univ., 157 Hammond Bldg., University Park, PA 16801)

Surface waves are waves with speeds somewhat less than the speed of sound in the adjoining fluid and which are largely carried by the fluid itself. Such waves cannot propagate on a flat plate at frequencies below the coincidence frequency because the imaginary part of the surface impedance has the wrong sign. The correct sign is when the surface has spring-like behavior rather than mass-like (inertial) behavior. Based on such an analogy, one would not expect a cylindrical shell to support a surface wave at very low frequencies. However, the shell has a squeezing stiffness that makes its behavior more like that of a plate on an elastic foundation with relatively weak spring constant. An analysis is developed that shows that this spring constant per unit area is Eh/R^2 , where h is the shell thickness and R is the shell radius. The dispersion relation

for the axial surface wave is developed and shown to give predictions consistent with recent experimental results and computational simulations of Williams, Houston, and Bucaro [J. Acoust. Soc. Am. **87**, 513–522 (1990)]. [Work supported by ONR and by the William E. Leonard endowment to Pennsylvania State University.]

1:45

7SA2. Uncoupling supersonic Lamb waves on plates and shells from the surrounding fluid: Approximate condition on the phase velocity. Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Lamb waves propagating with a phase velocity c_l along a fluid-loaded plate or shell generally leak or radiate wavefronts at an angle determined by the trace velocity matching condition when c_l exceeds the velocity of sound in the surrounding fluid. Ray representations of the backscattering by spherical and cylindrical shells show the leaky wave

contributions are proportional to the radiation damping β_i of the wave [S. G. Kargl and P. L. Marston, *J. Acoust. Soc. Am.* **89**, 2545–2558 (1991); N. H. Sun and P. L. Marston, *J. Acoust. Soc. Am.* (accepted for publication)]. A curious numerical result is that for certain leaky rays, the function $\beta_i(ka)$ can exhibit a minimum where β_i can be as small as 10^{-4} Np/rad and the associated resonances are very narrow. An approximate condition for such minima is evident by considering the particle displacement at the surface of a flat plate in a vacuum. It has been shown [D. C. Worlton, *J. Appl. Phys.* **32**, 962–971 (1961)] that there is no normal displacement of the plate's surface when the phase velocity of a Lamb wave equals the longitudinal wave speed c_L of the material. In the present work it is shown that the radiation damping exactly vanishes at this condition for a flat plate with an unbounded fluid on one or both sides. Normal displacements are essential for radiation by curved shells so that β_i should be minimized for $c_i \approx c_L$. This condition was numerically verified. The phenomenon allows for the supersonic transport of mechanical energy with negligible sound production. [Work supported by ONR.]

2:00

7SA3. Generalized Lamb waves near mode thresholds of spherical shells. D. H. Hughes and P. L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Pole trajectories as a function of frequency in the complex angular momentum plane for backscattering from empty spherical shells with and without significant fluid loading are presented. These are correlated with Lamb wave threshold features (that depend on Poisson's ratio) found on a flat plate in vacuum. [R. D. Mindlin, "Waves and Vibrations in Isotropic, Elastic Plates," in *Structural Mechanics* (Pergamon, New York, 1960); A. H. Meitzler, *J. Acoust. Soc. Am.* **38**, 835–842 (1965).] The trajectories for the fluid-loaded case are relevant to the high-frequency enhancements and longitudinal (or thickness) resonance of recent investigations [see, e.g., S. G. Kargl and P. L. Marston, *J. Acoust. Soc. Am.* **89**, 2545–2558 (1991)]. The usual poles lie in the first quadrant of the upper half-plane. Near the longitudinal resonance condition a pole associated with a symmetric wave is present in the second and fourth quadrants and is associated with the high-frequency enhancement. Its pole trajectory indicates that the phase and group velocity are oppositely directed over a small range of ka where the enhancement occurs. [Work supported by ONR.]

2:15

7SA4. Biologically inspired approaches for reduced order control of distributed elastic systems. James P. Carneal and Chris R. Fuller (Dept. of Mech. Eng., Virginia Polytech. Inst. and State Univ., Blacksburg, VA 24061)

The control of distributed systems by multiple piezoelectric actuators has shown much promise. However for higher modal densities, many actuators are required and the order of the controller (i.e., number of channels) becomes large, leading to implementation problems and performance degradation. In this paper, a control approach, which was largely inspired by biological systems, is analytically investigated so as to reduce the order of the main controller. The inspiration for the system was derived from biological muscle control, where one main signal from the brain is sent to a large area of muscle tissue, but is processed locally by cell action into multiple subsequent signals for individual muscle cell elongation or contraction. In the present study, the system to be controlled consists of a simply supported beam excited by a harmonic point force disturbance. Control is attempted by multiple piezoelectric actuators attached to the beam surface. One actuator is chosen as the master actuator and is under direction of the main controller. The other (slave) actuators derive their control inputs by localized learning rules related to the behavior of their neighbor actuators. The master actuator uses linear quadratic optimal control theory while very simple local learning rules are employed to minimize beam vibrational energy density. The results presented demonstrate that the use of

the multiple slave actuators in conjunction with the main channel of control significantly enhances control performance over the single actuator case, particularly for off resonance cases, by reducing control spillover. [Work supported by NASA Langley and ONR.]

2:30

7SA5. A method for simultaneously determining structural response and bistatic target strength of immersed structures. Joseph A. Clark and Michael A. Sartori (Carderock Div., Naval Surface Warfare Ctr., Bethesda, MD 20084-5000)

Structural response measurements can reveal the physical mechanisms that control scattering of acoustic waves by an immersed structure. Bistatic target strength measurements provide a means for fully evaluating effects of the physical mechanisms on far-field scattering. Phase-coherent, simultaneous measurements of both quantities would permit structural acoustics investigators to relate in detail scattering mechanisms with corresponding far-field scattering effects. The simultaneous measurement method to be described in this talk utilizes pressure measurements made with a two-layer conformal array to determine pressure and pressure gradient distributions on a measurement surface that is located in the evanescent near field of the immersed structure and conforms to its shape. The acoustic field is back-projected to the structure surface with a near-field-holography algorithm to determine structural response and is forward-propagated to the acoustic far field with an algorithm based on the exterior Helmholtz integral to determine bistatic target strength. The method is illustrated with a simulated study of scattering from an idealized cylinder. [Work supported by ONT.]

2:45

7SA6. Acoustic scattering from fluid-loaded plates with arbitrary boundary conditions. R. Kaina and J. M. Cuschieri (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

The dynamic response and acoustic scattering from a finite, fluid-loaded, rectangular plate, simply supported on two edges and with arbitrary boundary conditions on the remaining two edges are investigated. An analytic solution of the equation of motion of the plate is obtained by using a modal decomposition in the direction perpendicular to the simply supported edges and by using a spatial transform into the wave-number domain for the remaining direction. Since the plate is finite, when taking the spatial transform, four response-related parameters are obtained in the result of the transform. From knowledge of the boundary conditions a set of simultaneous equations can be set up and solved to evaluate these parameters. By obtaining the solution in the wave-number domain, the need for assuming a known mode shape is eliminated. Using this approach for solving the fluid-structure coupling problem, arbitrary boundary conditions can be considered. In this paper, the above approach is used to solve for the input and scattered acoustical power from two edge-coupled plates with a discontinuity at the common junction. [Work sponsored by ONR.]

3:00

7SA7. Acoustic wave scattering from thin shell structures. Andrew Norris and Naresh Vasudevan (Dept. of Mech. & Aerosp. Eng., Rutgers Univ., P.O. Box 909, Piscataway, NJ 08855-0909)

The response of heavily fluid-loaded thin shells in the mid-frequency regime is studied using a general asymptotic theory. The technique employs the ideas of matched asymptotic expansions and splits the total shell response into the outer or background component and the inner or resonant contribution. This formulation is applicable to thin shells with varying material and geometrical properties. The background field satisfies an impedance condition that does not correspond to either the so-called soft or rigid conditions. The approach is to expand both the

outer and inner fields in inverse powers of $1/kR$, where R is a typical radius of curvature of the shell. The leading term in these expansions is selected so as to reflect the dominance of the tangential motion near a compressional resonance and the stronger radial motions in the outer region. Also derived are uniform asymptotic representations. The asymptotic method is applied here to study scattering from various non-separable two-dimensional geometries. The numerical procedure is to first compute the outer solution, which requires solving for a particularly simple impedance condition on the surface. Then the inner resonances are added to give the total response. The numerical method employed in finding the outer solution is MOOT or the method of optimal truncation. Comparisons are made with the solution to the full set of coupled equations. [Work supported by ONR.]

3:15

7SA8. Near-field sound radiation from a line-driven fluid-loaded infinite flat plate with periodic and nonperiodic attached rib stiffeners. Benjamin A. Cray (Naval Undersea Warfare Ctr. Detachment, New London, CT 06320)

The near-field acoustic pressure radiated from a line-driven, fluid-loaded, rib-stiffened plate has been obtained using numerical integration techniques. The plate has been configured to have two sets of rib stiffeners, though the formulation given may be extended to include additional rib stiffener sets. The stiffeners composing a given set are identical and are spaced periodically with distance l . However, one set of stiffeners is shifted by an amount Δ from the other set. In this manner, portions of the plate may be configured with repeating sections having nonperiodic rib spacing. The stiffeners exert reactive forces upon the plate, but not angular moments. Results are presented that show the variation of near-field acoustic pressure along the surface of the plate at a given stand-off distance and at a fixed frequency. At certain frequencies, the investigation revealed that the wavelength of the stiffened plate's near-field acoustic oscillations—due to the propagating flexural wave—was much greater than that given by the unstiffened plate's bending wavelength. The effect may be due to the stiffened plate's flexural response being split into two components that are close together in wave number. The wavelength of acoustic oscillation appears to result from taking the difference of the two spectral components.

3:30

7SA9. Transient analytic and numerical results for the fluid-solid interaction of prolate spheroidal shells. Janet B. Jones-Oliveira (LLNL, P.O. Box 808, L-84, Livermore, CA 94551)

A transient solution is presented that models the fluid-solid interaction of a thin elastic prolate spheroidal shell loaded end-on by a nonconservative acoustic shock wave. Solutions to the Lagrangian equations of motion are provided for the normal and tangential shell displacement fields, as well as for the incident, scattered, and radiated fluid pressure fields. The temporally and spatially dependent shell displacement and fluid pressure fields are expressed modally in terms of an orthonormal basis that is selected specifically to satisfy the kinematic boundary conditions at the fluid-solid interface identically. Further, the modal representations for the shell displacements resolve the boundary conditions imposed by the problem configuration; and the modal representations for the fluid field satisfy causality. The solutions are developed in terms of prolate spheroidal angular and radial wave functions. For both analytic and computational reasons, the shell displacement and fluid pressure fields are ultimately expressed in terms of Legendre polynomials and modified spherical Bessel functions of the first and third kinds. The resulting solutions are exact within the limit of series solutions. A novel analytic/numerical hybrid solution technique is introduced for solving the inhomogeneous coupled PDE's; the spectral method is a Galerkin variation. The explicit analytic solutions converge to the exact solution of the actual coupled differential equations over the entire temporal and spatial domains both in the structure and in the fluid. Series truncation is postponed until the final expressions and their

mathematical structure are known, thereby ensuring that all relevant physics has been correctly modeled having taken full advantage of inherent symmetries. Numerical results for the free vibrations and for the transient fluid-solid interactions of a fluid-loaded prolate spheroidal shell are presented. Specifically, the axisymmetric *in-vacuo* nondimensionalized frequencies are tabulated for three characteristic cases corresponding to low, intermediate, and high aspect ratio geometries. The fluid loading is shown to shift the frequencies, as well as introduce additional structural frequencies. Plots of the shell displacements, velocities, accelerations, strains, and strain rates are presented for the submerged shock-loaded low aspect ratio case. Insights into the qualitative and quantitative effects of the fluid on the structural response are revealed. DOE-MACSYMA was used extensively to develop and verify the solutions to this problem. [This work was performed under the auspices of DOE by LLNL under Contract No. W-7405-Eng-48.]

3:45

7SA10. Asymptotically correct shell theories with fluid loading for the thin spherical shell. Cleon E. Dean and Michael F. Werby (Code 221, NRL, Stennis Space Center, MS 39529-5004)

So-called shell theories give reasonably good results for the motion of a bounded elastic shell by positing that various parts of the shell move together in some reasonable manner. They also can give physical insight into the motions of the shell while using less computational time and resources than exact elastodynamic calculations. The use of various assumptions about the motions and fluid loading of the thin spherical shell gives rise to several shell theories. The results derived from these theories are compared with the exact results with particular emphasis on the large size parameter (large ka) limit for the flexural and extensional Lamb modes. Limitations of each of the methods are then outlined as well as those of shell methods in general. [Work supported by ONR/NRL and by ONT Postdoctoral Fellowship Program.]

4:00

7SA11. Acoustic scattering from a fluid-loaded finite plate with an attached structural fuzzy. Daniel A. Russell and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., 117 Appl. Sci. Bldg., University Park, PA 16802)

In 1986 Christian Soize and his co-workers at Office National d'Etudes et de Recherches Aéropatiales in Chatillon, France introduced the concepts of fuzzy structures to account for missing degrees of freedom in structural vibration problems. The elements of Soize's methods have been recently outlined [J. Acoust. Soc. Am. **89**, 1867 (1991)]. The present paper concerns the scattering of an incident plane wave from a fluid-loaded finite plate to which a discrete structural fuzzy has been attached. Analyses will be presented for cases where the frequency of the incident field is both above and below the cutoff frequency of the fuzzy attachment. Below the cutoff frequency the fuzzy adds an additional fluctuating mass term to the *in-vacuo* modal masses that contribute to the plate velocity solution. Above the cutoff frequency, one must account not only for the mass contribution of the attached fuzzy, but also for the fluctuating internal damping and modal density of the fuzzy. [Work supported by ONR.]

4:15

7SA12. The acoustic radiation efficiency of metal and composite panels. Richard P. Szwerc (Code 1944, David Taylor Res. Ctr., Bethesda, MD 20084) and Richard M. Weyer (Penn State Univ., State College, PA 16804)

The acoustic radiation efficiency of solid metal and composite panels submerged in water has been analytically and experimentally determined. The experimental results validate the theoretical expressions of Blake [W. K. Blake, *Mechanics of Flow-Induced Sound and Vibration*,

Vol. I, 331-333 (1986)]. The dependence of the radiation efficiency upon the wave speed of the panels when excited by a point force has been observed, thus emphasizing the independence of the radiation efficiency and the structural damping of the panels.

4:30

7SA13. Effects of junctions on radiation by plate and shell modes into a surrounding fluid. I-Tai Lu, Henry L. Bertoni, and Yuqi Yao (Weber Res. Inst., Dept. of Elec. Eng., Polytech. Univ., Rte. 110, Farmingdale, NY 11735)

This discussion considers methods for computing radiation into surrounding water due to vibrating sources within a submerged vessel with internal structures. The vessel is viewed as a collection of plates and shells; the vibration of each component can be written as a superposition of flexural, extensional, and shear motions. Depending on the wave velocity, there can be two components to the radiated field. For example, the extensional wave can radiate continuously into water as it propagates along the outer hull. The angle of radiation is $\sin^{-1}(v_w/v_p)$, where v_p and v_w are the velocities of the extensional mode and the acoustic wave in water, respectively. In addition, there will be radiation over a 180° spectrum of angles that originates from the vicinity of junctions between the hull and internal structures such as ribs and bulkhead. This radiation is an end effect arising from the fact that the shell/plate waves have a finite starting point. (Note that the flexural mode *can* also radiate through this diffraction mechanism even when the frequency is below the coincidence frequency.) An important aspect of this paper will be to evaluate the relative importance of these two mechanisms for various shell/plate modes. [Work supported by ONR.]

4:45

7SA14. Acoustic scattering from a fluid-loaded thin plate/shell with a reinforcing rib. I-Tai Lu, Henry L. Bertoni, and Yuqi Yao (Weber Res. Inst., Dept. of Elec. Eng., Polytech. Univ., Rte. 110, Farmingdale, NY 11735)

Acoustic waves incident from water are partially reflected at the

outer shell of a submerged vessel. In addition, they excite elastic vibrations of the internal structure of the vessel, which in turn re-radiate into the water. Thus, predicting the scattered acoustic field requires the ability to model and account for the internal structures. This discussion considers methods for computing scattering from a flat plate or a cylindrical shell with a reinforced rib. Full elastic properties of the rib are taken into consideration. The leaky modes excited by incident waves are coupled to other modes in the fluid-loaded plate/shell as well as in the rib. The mode coupling coefficients at the rib are highly dispersive because waves reverberate between its free edge and the junction where the rib is joined to the plate/shell element. These plate/shell waves induced by the rib can re-radiate to the water through two mechanisms: leakage and diffraction (see the preceding companion paper). The effects of the rib on the scattered field will be examined in detail. [Work supported by ONR.]

5:00

7SA15. Low-frequency pressure wave propagation in liquid-filled, flexible tubes. Cato Bjelland (GECO-PRAKLA, FI, N-5100, Isdalstø, Norway) and Leif Bjørnø (Tech. Univ. of Denmark, Lyngby, Denmark)

A model has been developed for propagation of low-frequency pressure waves in viscoelastic tubes with distensibility of greater importance than compressibility of the liquid. The dispersion and attenuation are shown to be strongly dependent on the viscoelastic properties of the tube wall. The complex, frequency-dependent moduli of relevant tube materials have been measured in a series of experiments using three different experimental procedures, and the data obtained are compared. The three procedures were: (1) ultrasonic wave propagation, (2) transverse resonance in bar samples, and (3) moduli determined by stress wave transfer function measurements in simple extension experiments. The moduli are used in the model to produce realistic dispersion relations and frequency dependent attenuation. Signal transfer functions between positions in the liquid-filled tube can be synthesized from the model and are compared with results of experimental pressure wave propagation in the liquid-filled, flexible tube. A good agreement between experimental data and theoretical predictions is found.

THURSDAY AFTERNOON, 14 MAY 1992

SALON I, 1:00 TO 4:20 P.M.

Session 7SP

Speech Communication: Speech Processing: Automatic Recognition, Analysis, and Synthesis

David R. Williams, Chair

Sensimetrics Corporation, 64 Sydney Street, Cambridge, Massachusetts 02139

Chair's Introduction—1:00

Contributed Papers

1:05

7SP1. Lattice hidden Markov models for speech recognition. Lu Chang and M. M. Bayoumi (Dept. of Elec. Eng., Queen's Univ., Kingston, Ontario K7L 3N6, Canada)

The hidden Markov model (HMM) has been successfully applied to the area of automatic speech recognition. However, HMM has two assumptions, namely, Markovian transition of its underlying states and time-independent emissions of observations. As is widely known, speech signals are highly correlated. And for speech recognition, temporal in-

formation is very important. A lattice HMM (LHMM) is proposed to address the difficulty of conventional HMM. The new model has two sequences, a hidden state sequence and an observable group sequence. Both sequences are assumed to have a Markovian property. However, the transitions of these two sequences are dependent on each other. The observations of the model are functions of the state and the group sequence. It should be noted that although the group sequence is observable, it is represented as a transformation of the observation sequence. The re-estimation formula for the model will be presented. The training of LHMM needs some care. Segmented data are clustered using DTW to find the typical group transition paths. The group transition matrix is estimated from the transition paths. This procedure is devised to increase the discrimination power of the model.

1:20

7SP2. Feature extraction from a cochlear model representation: A novel supervised/unsupervised neural network hybrid for speech recognition. Gary N. Tajchman (Dept. of Cognitive and Linguistic Sci., Brown Univ., Box 1978, Providence, RI 02912) and Nathan Intrator (Ctr. for Neural Sci., Brown Univ., Box 1843, Providence, RI 02912)

A novel supervised/unsupervised hybrid neural network training algorithm [Intrator, "Combining exploratory projection pursuit with projection pursuit regression," preprint (1992)] is applied to a speech recognition task. The input representation for the neural network is produced by Richard Lyon's cochlear model as implemented by Slaney [M. Slaney, "Lyon's Cochlear Model," Apple Tech. Report #13, Apple Comput., Inc., Cupertino, CA 95014]. This detailed, high-dimensional representation is projected through the units of the hidden layer of a back-propagation-like architecture into a much lower dimensional space (spanned by the number of hidden units). This space is constructed by optimizing a combination of the familiar MSE minimization and an unsupervised measure of the goodness of the projections. The goodness measure is based on the multimodality of the projected distributions. Both constraints are powerful and useful when used alone. However, when they are combined, the resulting network has the potential to learn and generalize much more robustly than either alone. This technique was applied to the classification of 16 classes of stressed vowels extracted from the TIMIT Acoustic-Phonetic Continuous Speech Corpus [NIST Speech Disc 1-1.1 (October 1990)]. Comparisons between the hybrid method and plain back propagation are discussed.

1:35

7SP3. Efficient representation of LPC excitation using nonuniform frequency-domain sampling. Sunil K. Gupta and Bishnu S. Atal (Speech Res. Dept., AT&T Bell Labs., Murray Hill, NJ 07974)

An efficient representation of the LPC excitation is essential in predictive coding systems for synthesizing high-quality speech at low bit rates. In this paper, a method is presented that takes advantage of the nonuniform spacing of auditory critical bands to achieve an efficient frequency-domain representation of LPC excitation. A segment of LPC excitation with a duration of N samples, represented as a Fourier series, requires $N/2$ sinusoidal components uniformly spaced along the frequency axis for exact reproduction of the excitation. Thus, for speech bandlimited to 4 kHz, a 10-ms segment requires 40 frequency components for exact reproduction. It was found that, by using uniform frequency spacing below 1 kHz and logarithmic spacing above 1 kHz, the number of sinusoidal components can be reduced to 15 without introducing any audible distortion in the synthetic speech signal. Subjective tests were conducted to determine the effective signal-to-noise ratio of synthetic speech for different numbers of sinusoidal components. These results and a tape demonstration of synthetic speech will be presented at the meeting.

1:50

7SP4. Control of Klatt speech synthesizer with high-level parameters. Corine A. Bickley, Kenneth N. Stevens, and David R. Williams (Sensimetrics Corp., 64 Sidney St., Cambridge, MA 02139)

This paper describes a system for synthesis of speech with a terminal analog synthesizer like the Klatt synthesizer. The control of the synthesizer is accomplished with a small number (about 10) of high-level (HL) parameters that capture directly certain attributes of the state of the vocal tract during speech production. The multiplicity of low-level parameters in the synthesizer are derived from the HL parameters through a set of mapping relations. The HL parameters consist of (1) the fundamental frequency; (2) a specification of the vocal tract in terms of the first three or four natural frequencies of the tract assuming no opening to the glottis or the nose; (3) the areas of three major orifices in the vocal tract: the glottal opening, the velopharyngeal opening, and (for consonants) the principal constriction in the vocal tract; (4) a measure of active expansion or contraction of the vocal-tract volume for obstruents; (5) a measure of the efficiency of turbulence noise generation for obstruents. The mapping relations for obstruents involve the calculation of airflows, pressures, wall displacements, glottal waveform characteristics, vocal-tract losses, and noise due to turbulence. Some examples of the mapping relations for both obstruents and sonorants are given. [Research supported by the National Institutes of Health.]

2:05

7SP5. Examples of speech synthesis with high-level parameters controlling a Klatt synthesizer. David R. Williams and Corine A. Bickley (Sensimetrics Corp., 64 Sidney St., Cambridge, MA 02139)

This paper describes the synthesis of a number of simple syllables with a Klatt synthesizer that is augmented to include the capability of controlling ten high-level (HL) parameters. These HL parameters are mapped into a much larger number of low-level (LL) parameters in a way that incorporates constraints on these parameters imposed by the acoustics and aerodynamics of the speech production system. Examples are presented to show that the synthesis of a pair of utterances that differ in one feature (such as /pa, ba/, or /ba, ma/, or /ba, da/) can be realized by keeping most HL parameters the same and changing only one or two parameters (such as glottal opening, velopharyngeal opening, or trajectories of one or two formants). Although a complex pattern of LL parameters with precise temporal synchrony is required to synthesize these utterances, the pattern of HL parameters is relatively simple, and the timing relations are often rather lax. Comparisons of the parameter sets needed to synthesize a variety of consonants reveal how HL synthesis captures generalizations about speech sound classes that traditional acoustic synthesis does not. [Research supported by the National Institutes of Health.]

2:20

7SP6. Influence of an internal reference system and cross-modality matching the subjective rating of speech synthesizers. Chaslav V. Pavlovic,^{a)} Mario Rossi, and Robert Espesser (LA 261, CNRS, Institut de Phonétique, Univ. de Provence, 29 Av. Robert Schuman, 13621 Aix en Provence, France)

In previous studies it was concluded that contextual invariance and subject invariance of categorical and magnitude estimates of speech quality could be improved by introducing a reference system and by normalizing the results with respect to it. The reference signal used in the previous studies was natural speech. The use of such a reference system may present problems for applications where cross-language comparisons of synthesizers are made. In particular, this refers to the difficulty of ensuring equal subjective quality of different talkers in different languages. In this study the possibility of substituting an actual reference signal with an "internal" reference defined to the subject as

the system of optimal quality is investigated. Another objective of this study is to explore whether a sometimes difficult task of free number production required in magnitude estimations could be replaced by cross-modality matches using lines of various lengths produced by subjects on a computer screen. The main concern here was related to the

unknown effects of the limited width of the computer screen on the magnitude estimation task. [This research was made possible by grant No. 2589 from the EEC Espirit SAM project.]^{a)} Also at Univ. of Iowa City, IA.

2:35–2:50

Break

2:50

7SP7. An unsupervised method for learning to track tongue position from an acoustic signal. John Hogden, Philip Rubin, and Elliot Saltzman (Haskins Labs., 270 Crown St., New Haven, CT 06511)

A procedure for learning to recover the relative positions of the articulators from speech signals is demonstrated. The algorithm learns without supervision, that is, it does not require information about which articulator configurations created the acoustic signals in the training set. The procedure consists of vector quantizing short time windows of a speech signal, then using multidimensional scaling to represent quantization codes that were temporally close in the encoded speech signal by nearby points in a *continuity map*. Since temporally close sounds must have been produced by similar articulator configurations, sounds which were produced by similar articulator positions should be represented close to each other in the continuity map. Using an articulatory speech synthesizer to produce acoustic signals from known articulator positions, relative articulator positions were estimated from synthesized acoustic signals and compared to the synthesizer's actual articulator positions. High rank-order correlations, ranging from 0.92 to 0.99, were found between the estimated and actual articulator positions. Reasonable estimates of relative articulator positions were made using 32 categories of sound, and the accuracy improved when more sound categories were used.

3:05

7SP8. Description of contextual factors affecting vowel duration. Jan P. H. van Santen (AT&T Bell Labs., 2D-452, 600 Mountain Ave., P.O. Box 636, Murray Hill, NJ 07974-0636)

As an initial phase of a project on duration models for predicting segmental durations from contextual factors, two natural speech databases produced by a male and a female speaker with more than 50 000 manually measured segmental durations were analyzed. This large quantity of data made it possible to perform a detailed analysis of the effects of several contextual factors, including lexical stress, word accent, the identities of adjacent segments, the syllabic structure of a word, and proximity to a syntactic boundary. Among the key results were the following. (1) The contextual factors accounted for up to 90% of the variance, and reduced the within-vowel standard deviation by a factor of 3. (2) There were complex interactions between factors, in particular between boundary proximity and postvocalic consonant identity and between lexical stress and syllabic word structure. (3) The effects of adjacent segments were reducible to the effects of voicing and manner of production; effects of place of articulation were negligible. (4) Proximity to a boundary should be measured in terms of syllabic and segmental position, not in terms of the sum of the intrinsic durations of segments between the target and the boundary. The results were compared with data reported by Crystal and House [J. Acoust. Soc. Am. 83, 1551–1573 and 1574–1585 (1988)].

3:20

7SP9. Improved vowel recognition using the discrete cosine transform. D. J. Burr (Comput. Systems Res. Dept., Bellcore, 445 South St., Morristown, NJ 07960)

Some experiments are described that use a discrete cosine transform (DCT) to represent vowel spectra for classification by a neural network. The recognition accuracy of a neural network trained by back propagation was compared to that of a simple Gaussian classifier. Twelve English vowel categories from the TIMIT database were used [aa, ae, ah, ao, ax, eh, er, ih, ix, iy, uh, uw]. Training was done in a speaker-dependent manner using 152 male speakers from all geographic regions. Eight 16-ms DFT spectra were computed at 8-ms intervals about the center of each vowel. Sixteen DCT coefficients were derived from the 250- to 4250-Hz interval in each 8-kHz spectrum. Average DCT and delta DCT vectors over the eight frames were used as input features. Additional features included the first six peak frequencies of the DCT spectrum and two pitch parameters from the DCT coefficients. Best performance was obtained by the neural network with all 40 input features, resulting in 58.2% recognition accuracy. This compares favorably to a cochleagram representation using the same vowel classes [Muthusamy *et al.*, ICASSP-90]. The DCT also appears to require fewer coefficients than an equivalent cepstrum-based vowel classifier [Burr, ICASSP-92].

3:35

7SP10. Automatic measurement of intonation. Gerard W. G. Spaai, Arent Storm, and Dik J. Hermes (Inst. for Perception Res./IPO, P.O. Box 513, NL 5600 MB, Eindhoven, The Netherlands)

If speech intonation is only represented by an unprocessed series of pitch measurements, the interpretation can be hampered by three factors. First, because of the presence of unvoiced parts in an utterance, the continuously perceived pitch contour is disturbed by interruptions. Second, in many cases, speech is characterized by involuntary pitch perturbations that either cannot be heard at all, or do not contribute to the perception of intonation. Third, the perceptual meaning of a pitch movement depends upon its position within the syllable, in many cases with respect to the vowel onset. A correct interpretation of a pitch contour, therefore, requires the position of the vowel onsets to be known, too. These problems can be solved by interpolating the pitch at unvoiced parts from the adjacent pitch measurements, by removing all perceptually irrelevant details from the contour, and by indicating the vowel onsets. Besides presenting the procedures, a system will be presented which performs these tasks in real time, and which is currently used in an evaluation of its usefulness in teaching intonation to deaf persons. The applicability for the use of this intonation meter in training intonation of foreign languages will be indicated. [Work supported by Instituut voor Doven, St.-Michielsgestel, Netherlands.]

7SP11. A new technique for formant extraction by adaptive smoothing of Fourier spectra. Venkatesh R. Chari and Carol Y. Espy-Wilson (Dept. of Elec., Comput., and Systems Eng., Boston Univ., 44 Cummington St., Boston, MA 02215)

A new technique to automatically track formant frequencies in voiced speech was developed. Continuity constraints were imposed by adaptively smoothing the short-time Fourier representation. At each point in the time-frequency field, the average energy distribution within a linear window was computed as a function of angular displacement. The angle at which the energy density was closest in magnitude to the point under consideration was chosen as the most likely direction of energy correlation. Weighted smoothing was then performed in that direction. This allowed potential candidate peaks to be enhanced if they lay on a formant track but were not robust enough to be picked on their own merit. Continuous and distinct peak tracks were thus generated even in cases where formants were close. Higher level speech knowledge sources were then called upon to assign peak tracks to formant slots. A measure of reliability was also estimated for each assignment. Experiments conducted on sentences from the TIMIT database yielded very

encouraging results. Results will be presented. [Work supported by NSF.]

4:05

7SP12. Phoneme recognition system. James Lisonbee, Dan Tebbs, and Paul Wheeler (Dept. of Elec. Eng., Utah State Univ., Logan, UT 84322-4100)

The possibility of phoneme recognition using a hardware-based spectral sampling filter and a 386- μ processor software-based neural network to accurately recognize ten spoken vowel sounds of the English language was explored. To apply the neural network to speech recognition, a set of inputs conducive to the neural network architecture must be supplied. The spectral sampling filter does this by creating a pattern representing energy content in the first and second formants. This analog pattern is converted to digital and communicated to the neural network. The neural network was trained using a back-propagation technique. The neural network sees the localized energy changes across time as a distinct pattern for each phoneme. The neural network executes a recognition sequence and produces an integer correlating to the spoken phoneme.

THURSDAY AFTERNOON, 14 MAY 1992

SALON D, 12:40 TO 5:15 P.M.

Session 7UW

Underwater Acoustics: Signal Processing

Ronald A. Wagstaff, Chair

Naval Research Laboratory, Code 245, Stennis Space Center, Mississippi 39529-5004

Chair's Introduction—12:40

Contributed Papers

12:45

7UW1. Phase-matching filter techniques for low signal-to-noise data. G. J. Orris, B. E. McDonald, and W. A. Kuperman (Naval Res. Lab., Washington, DC 20375-5000)

A digital filter is being developed for use in low signal-to-noise situations where the shape of the noise spectrum is known *a priori*. Candidate noise signals are generated and subtracted from "observed" (real or idealized) pressure-time series by minimizing total power in the output with respect to the phases of the noise components. The minimization process results in matching the phase of each candidate noise component to that of the input signal at the same frequency. When the noise spectrum is sufficiently smooth and distinct in shape from that of the signal, good results have been obtained for idealized cases involving signal-to-noise ratios of -80 to -100 dB. At these levels there are no identifiable peaks in the power spectrum to indicate even the existence of a signal. With no *a priori* knowledge of the phase of either the signal or noise, an almost undistorted signal can be recovered. In more realistic cases involving recorded noise, however, filter performance is degraded. Methods to improve filter performance in nonidealized cases are under investigation.

1:00

7UW2. Temporal arrival structure from matched-filter processing of an LFM signal across two convergence zones using Difar sonobuoys. Thomas N. Lawrence, Kirk L. Holub, and Nancy R. Bedford (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713)

An experiment conducted near the Nares Abyssal Plain broadcast LFM and cw signals to investigate propagation across two convergence zones (CZ). Air-launched Difar sonobuoys were placed in the first and second CZs simultaneously, and cw bursts of about 1000 Hz, and LFMs covering 900 to 1100 Hz were recorded. A time-delayed echo repeater, which rebroadcast received signals omnidirectionally, was located in the first CZ. These signals were recorded via an expanded videocassette recording system [S. G. Payne, N. R. Bedford, and L. A. Thompson, "The AEAS Poco Amigo Airborne Acoustic Measurement System," ARL-TR-88-28], which demultiplexes signals from each Difar sonobuoy into its omnidirectional, north-south, and east-west components. Matched-filter processing was applied to these data to obtain a temporal arrival structure. In the case of the second CZ, all three components of the Difar buoy were separately recorded thus allowing horizontal directionality via beamforming. Using calibration information from directional hydrophones, vertical arrival angles were determined

[M. B. Bennett, L. A. Thompson, and P. T. Eisman, "Moored Data Buoy for Shallow Water Acoustic Studies" (in preparation)]. The arrival structure of the primary source and the echo repeater will be contrasted. [This work is supported by the ASW Environmental Acoustic Support Program.]

1:15

7UW3. Sensitivity of the deconvolution of ocean transients to environmental mismatch. Michael K. Broadhead, Robert L. Field, and James H. Leclere (NRL, Stennis Space Center, MS 39529)

The single-channel deconvolution of acoustic transients for source classification purposes can be sensitive to small errors in the computed ocean Green's functions. These errors arise due to incomplete environmental and source location information. The sensitivity was studied by perturbing the Green's functions in such a way as to produce improved source estimates from field measurements. A systematic technique to do this was developed using a weighted, linear optimization approach. It was discovered that small changes in the computed Green's functions could improve the source estimate significantly. It was also determined that the match between the simulations and the measured data is closely connected to the match between the source estimates and the measured source waveform, hence the sensitivity is not just due to deconvolution. This relationship was quantified by using the correlation coefficient. This study is part of an effort to determine the applicability of deterministic deconvolution to the problem of transient classification. [Work supported by ONR under NRL program management.]

1:30

7UW4. Pulse compression using passive phase-conjugate processing. David R. Dowling and Darrell R. Jackson (Appl. Phys. Lab., College of Ocean and Fishery Sciences, Univ. of Washington, Seattle, WA 98105)

Passive phase-conjugate (PPC) processing consists of using the first arrival, from a stream of pulses that have traversed a refractive medium, as the matched filter for later pulse arrivals. Its connection with true "active" phase conjugation [see Jackson and Dowling, *J. Acoust. Soc. Am.* 89, 171 (1991)] lies in invoking reciprocity for the back propagation. When the intervening acoustic medium changes slowly, the processor creates unambiguous maxima regardless of the number of propagation paths between the source and the receiver. The technique can be easily extended to receiving arrays by coherently summing the PPC output from each hydrophone. Differences in the height and phase of the peaks in the PPC processor output from different pulses are produced by changes in the acoustic medium and the source-array geometry. The results from recent acoustic propagation experiments, including IWAC '90, will be used to illustrate some of the benefits of this technique. Applications will also be discussed. [Work supported by NRL.]

1:45

7UW5. Optimization of acoustic array design using the simulated annealing algorithm. Thomas J. Hayward (Naval Res. Lab., Washington, DC 20375-5000)

The simulated annealing algorithm [S. Kirkpatrick *et al.*, *Science* (May 1983)] has been applied recently to a number of underwater acoustics problems. In this paper methods developed in recent years for the optimization of sonar receiver array configurations, including both horizontal and vertical arrays, are reviewed. Optimization of horizontal planar-array configuration to minimize noise gain in a 2-D-isotropic noise field results in highly ordered configurations resembling crystalline or quasicrystalline structures. Vertical-array configuration optimization for conventional beamforming is based on maximization of signal-to-noise ratio at the array output. The optimized vertical-array configura-

tions exhibit far less regularity of spacing, due to the complexity of the acoustic field in the vertical. Nevertheless, the results are readily understood as a trade-off between placement of sensors at depths with high signal-to-noise ratio and separation of sensors to reduce the array noise gain. Optimization for matched-field processing is also considered.

2:00

7UW6. The learning capabilities of an artificial neural network for classifying acoustic signatures. Fred C. DeMetz (P.O. Box 28267, Panama City, FL 32411-8267)

The ability of an artificial neural network to learn to identify acoustic sources from a digital series of their spectral or temporal features has been studied experimentally. The learning system was comprised of a feedforward network employing a backward propagating delta rule for error correction. A three-layered network configuration of 30 input units, 15 hidden units, and 1 output unit was utilized. The network was first trained with patterns of 30 line spectra, some containing source lines and some containing noise only. Then the ability of the trained network to correctly identify new patterns of the source and noise data was tested. The results indicated several interesting characteristics relating to the utility of neural networks in the classification of acoustic waveforms. The network achieved classification accuracies exceeding 95%, with low false alarm rates, for frequency stable, high signal-to-noise spectra. For low signal-to-noise spectra, with fluctuations on both the frequency and amplitude of the source lines, the network tended to perform poorly.

2:15

7UW7. The potential applications of fuzzy logic to problems in underwater acoustics. Marc C. Leonetti (Comput. Sci. Corp., P.O. Box N, Moorestown, NJ 08057)

The methods of fuzzy logic and fuzzy statistics are applied to problems in underwater acoustics. Specifically, problems in underwater acoustics that require a decision to be made from incomplete and/or conflicting evidence are viewed from the perspective of fuzzy engineering. A method for making decisions based on the fusion of probabilities of multisensor inputs is discussed, and the role of a fusion rule is examined.

2:30

7UW8. Three-tier computerized system using ambient noise to assess the performance of complex multielement sonar systems. R. A. Wagstaff (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Experience with both developmental and Fleet operational sonar systems suggests that it is not uncommon for a sonar to be operating far below its design performance level without the sonar operator's knowledge that the performance is degraded. The magnitude of the degradation is often far more than an increase in gain that might be expected from a new generation improvement in that sonar. Hence, it makes sense to monitor the performance of the sonar system to detect degraded performance, to identify problems or faults when they exist, and to know when to correct the faults if it is possible to do so. Furthermore, the sonar is generally essential to an ASW mission; an assessment of its performance must be thorough and rapid. Such testing could take days, if done manually by a sonar technician. However, the task can be done more thoroughly, and in as little time as 2 or 3 min, by a smart computer program that does the analysis and decision making and uses ambient noise as the noise source. This paper discusses a three-tier comprehensive system of computer-based sonar system performance monitoring and data quality assessment algorithms. The algorithms and analysis techniques are discussed, and the results obtained on recent sea tests with vertical line arrays and towed horizontal line arrays are presented. In some cases, the sonar systems were shown to be operating at

a small percentage of their theoretical or expected levels. Furthermore, if the degraded states had been known in real time, simple fixes in the field could have returned the performance of some of those systems to nearly theoretical.

2:45

7UW9. Active acoustic classification of underwater targets based on time-frequency analyses. Roger H. Hackman, H. Arnold, and J. Reagan (Lockheed Palo Alto Res. Labs., Palo Alto, CA 94304-1191)

Active acoustic classification of underwater targets is an important problem of current interest. Previously published results have relied on

the spectrum of the echo return as an input to a neural net classifier. In this paper, the echo return is regarded as a classic example of a nonstationary signal; the representation/analysis of such signals is normally accomplished by time-frequency distributions. The potential of a time-frequency representation of the target echo for classification purposes is investigated. The echo returns for a class of six targets (three spheres, two finite cylinders, and a prolate spheroidal shell) are numerically synthesized from *T*-matrix solutions and corrupted with white Gaussian noise to serve as a basis for the classification study. Several feature extraction schemes are presented. The performance of a probabilistic neural net classifier is investigated as a function of SNR for the different feature vectors. Optimum performance is obtained when the feature vectors are based on scattering dynamics.

3:00-3:15

Break

3:15

7UW10. Active acoustic classification of underwater targets based on scattering resonances. Roger H. Hackman, Theagenis J. Abatzoglou, Hal Arnold, D. Specht, and T. Washburne (Lockheed Palo Alto Res. Labs., Palo Alto, CA 94304-1191)

This paper concerns the neural network based active acoustic classification of underwater targets. The feature vectors utilized in the training of the network are based on the scattering resonance parameters extracted from the echo return. It is expected that these parameters constitute close to an "optimum" set of feature vectors in that they greatly reduce the dimensionality of the echo return while preserving the "unique" aspects related to target identity. The classification performance of a probabilistic neural network classifier is evaluated for several different SNR levels for both monostatic and bistatic scattering configurations. The performance of several different resonance extraction algorithms (including the constrained total least squares technique and the SVD "Prony") is evaluated in varying levels of SNR. The classification study is based on echo returns synthesized from *T*-matrix solutions for five elastic targets (three spheres and two finite cylinders).

3:30

7UW11. Robust adaptive processing for underwater acoustic arrays. Russell C. Burkhardt (Code 5032, Naval Air Warfare Ctr.—Aircraft Div., Warminster, PA 18974) and Leon H. Sibul (Appl. Res. Lab., Penn State Univ.)

Adaptive processors for underwater acoustic arrays are known to be sensitive to uncertainty in the spatial structure of the signal. The performance of an adaptive algorithm is also degraded in cases where the second-order noise statistics cannot be accurately estimated such as when an array with many elements is involved and/or when the data are highly nonstationary. The conventional approach to uncertainty in the signal structure is to use constraints. These constraints, in general, do not directly use *a priori* knowledge of the uncertainty in the signal and also ignore the existence of any error in the noise statistics, the presence of which tends to increase sensitivity to other errors. In this work, an approach to this problem is developed in which realistic models of the uncertainties in both the signal and noise structures are used to derive robust adaptive algorithms based upon a minimax criterion. The realistic signal uncertainty models are also used to formulate robust matched-field processors. In addition, a transformation to mode space

based upon a minimax criterion for uncertainty in the characteristics of the acoustic channel is used to derive a robust matched mode processor.

3:45

7UW12. On the gain available from passive acoustic synthetic aperture processing. Edmund J. Sullivan (Code 103, Naval Underwater Systems Ctr., Newport, RI 02841) and William M. Carey (Code 302, Naval Underwater Systems Ctr., New London, CT 06320)

Since passive synthetic aperture processing (PSAP) has long been understood to involve a trade-off between spatial and temporal gain, it has often been stated that there is no "new gain" to be realized. The argument, based on the model of a plane wave in isotropic noise, is as follows. For a fixed time available for processing, PSAP reduces the amount of integration time available for each sensor while increasing the number of sensors available. Thus, temporal gain is reduced while spatial gain is increased by an equal amount. However, when the noise field is nonisotropic, a different result obtains. It is shown that in the case of fields containing localized noise sources, PSAP can provide an increase in overall gain due to the ability to beamform on a desired source while rejecting undesired sources. A generalized formulation of the spatial-temporal processing gain provided by PSAP in terms of the ratio of coherent noise to incoherent noise power is presented that demonstrates the gains achievable in such a nonisotropic noise field. It is further shown that, even in the case of an isotropic noise field, if sufficient signal to noise is available, an improvement in bearing resolution can be achieved.

4:00

7UW13. Acoustic pulse spreading and distortion due to frequency-dependent attenuation in water at high frequencies. J. B. Cole (U.S. Naval Res. Lab., Code 5126, Washington, DC 20375-5000)

For high-resolution imaging sonar, short high-frequency pulses are needed to reconstruct surface features. For example, 1-cm range resolution or better requires pulse durations of 12 μ s or less at frequencies of 1 MHz or greater. Because large bandwidths are required to synthesize the pulses, the signals spread in the time domain and distort as they propagate through the water due to frequency-dependent attenuation that rises as the square of frequency. These effects were studied for

various combinations of signal parameters, and the results are related to signal design considerations for high-resolution sonar ranging and imaging.

4:15

7UW14. Nonorthogonal signal decomposition. Nai-chyuan Yen (Phys. Acoust. Branch, Naval Res. Lab., Washington, DC 20375-5000)

To gain physical interpretations and insights from observed phenomena, it is often desirable to decompose a given function, i.e., the observed signal, into a set of nonorthogonal functions. This paper describes a signal processing algorithm based on the minimization of the mean-square error formulation of the decomposition analysis. The computation scheme is derived from a recursive gradient descent method to approach the correct answer for a set of over-determined simultaneous equations. The traditional matrix inversion technique is usually not workable for numerical operation because the matrices are often ill-conditioned. This suggested approach not only alleviates the computation problem, but also can more easily be implemented in a parallel architecture computer or a neural-like network. The signal received from a synthetic circular array [N. Yen, IEEE/JOE, Jan. 1992, pp. 40-47] is used, as a practical example, to illustrate that a composite signal can be reduced to a combination of the corresponding signals from various directions. Comparison of the results obtained by this algorithm with the traditional beamforming method supports the use of this new signal processing technique.

4:30

7UW15. Evaluation high-resolution frequency estimators for the determination of the eigenvalues of modes in a shallow water waveguide. Subramaniam D. Rajan and Saurav Bhatta (Dept. of Appl.

Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

In shallow water, the acoustic field due to a point source can be modeled as a sum of contributions from trapped modes propagating in the waveguide. The performances of two high-resolution methods (MUSIC and ESPRIT) in estimating the eigenvalues of the modes and in resolving closely spaced modes are studied using noisy synthetic data and field measurements. Simulation performed to investigate the effect of modeling errors indicate, that for a weakly range-dependent medium the performance of the algorithms is not adversely affected. However, errors in ranging and presence of colored noise in data have considerable impact on the performance of these algorithms. [Work supported by ONR.]

4:45

7UW16. Extracting ocean sound-speed information from acoustic ambient noise measurements using vertical hydrophone arrays in shallow water. David H. Berman (Dept. of Phys. and Astron., Univ. of Iowa, Iowa City, IA 52242) and Stephen N. Wolf (Naval Res. Lab., Washington, DC 20375)

The ocean sound-speed profile near a vertical hydrophone array in shallow water may be determined from measurements of the acoustic field due to distant noise sources. Sound speeds are obtained from second spatial derivatives of the noise covariance matrix. The distance to, and the bandwidth of, the noise generator and the geometric dispersion of the duct must yield a covariance matrix that is an incoherent sum of the acoustic modal components. The array need not span the water column; however, sound speeds are determined only at hydrophone locations. This technique may be useful for monitoring sound-speed changes in situations in which it would not be feasible to deploy or monitor independent sound-speed sensors near the acoustic array. Computer simulations show that sound-speed estimates are insensitive to noise that is uncorrelated between hydrophones. [Work supported by the Office of Naval Technology, Code 234.]

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, Chairman S3
 4706 Connell Drive, Raleigh, North Carolina 27612

H. E. von Gierke, Chairman, U.S. Technical Advisory Group (TAG) for ISO/TC43 and ISO/TC 108/SC4
 1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor for IEC/TC 29
 National Institute of Standards and Technology (NIST), Bldg. 233, Rm. 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, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock will be discussed. The Chairs of the U.S. TAGs for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelnitsky), and ISO/TC 108/SC4 (H. E. von Gierke) will report on current activities of these Technical Committees and Subcommittees. Reports will be given on the last meetings of ISO/TC 43 (in Australia from 5–12 December 1991) and of IEC/TC 29 (in New Zealand, from 25–29 November 1991).

THURSDAY AFTERNOON, 14 MAY 1992

SALON H, 3:00 P.M.

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with the

U.S. Technical Advisory Group Meetings for ISO/TC 43 Acoustics, and IEC/TC 29 Electroacoustics

G. S. K. Wong, Chairman S1
 Institute for National Measurement Standards (INMS), National Research Council, Ottawa, ON K1A 0R6, Canada

H. E. von Gierke, Chairman, U.S. Technical Advisory Group (TAG) for ISO/TC43
 1325 Meadow Lane, Yellow Springs, Ohio 45387

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 National Institute of Standards and Technology (NIST), Bldg. 233, Rm. A149, Gaithersburg, Maryland 20899

Standards Committee S1 on Acoustics. Working group chairs will report on their progress in the 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 last meetings of ISO/TC43 (in Australia from 5–12 December 1991) and of IEC/TC 29 (in New Zealand, from 25–29 November 1991).

Session 8MU

Musical Acoustics: Physical Principles and Performances

William J. Strong, Chair

Department of Physics and Astronomy, Brigham Young University, Provo, Utah 84602

Chair's Introduction—9:00

Invited Papers

9:05

8MU1. The cornett: Acoustics and construction. John R. McCann (Cornetts/Zinkenbau, 10351 S. 2505 E., Sandy, UT 84092) and William G. Mathews (Lick Observatory, Univ. of California, Santa Cruz, CA 95064)

Although the cornett was one of the most widely used wind instruments during the late 16th and early 17th centuries, the construction of replica cornetts at modern pitch has not been straightforward. The cornett is excited by a lip reed as in modern brass instruments but has an approximately biconical wooden bore with toneholes. Cornetts surviving in European collections generally play at pitch levels higher than the modern standard and many of the few extant mouthpieces are of uncertain, possibly more recent, origin. As a guide to constructing replicas at modern pitch, the input and distributed impedances have been computed in the frequency domain. The accuracy of the computed frequencies for each fingering combination, although encouraging, is at present insufficient for reliable *ab initio* construction. However, the computations can be useful in determining appropriate perturbations in the size and placement of toneholes or in the bore required to achieve small differential frequency changes.

9:35

8MU2. Woodwind models incorporating nonlinear air-column losses. Douglas H. Keefe (School of Music, DN-10, Univ. of Washington, Seattle, WA 98195)

The sound level within woodwinds under playing conditions is sufficiently high so that nonlinear losses are significant, yet time-domain models of sound production have assumed that the air column responds linearly. Using a linear air-column model, time-domain simulations of saxophone tones give results qualitatively similar to performed tones for low-register tones, but dissimilar results for second-register tones and multiphonics, both of which employ an open register hole. A time-domain analysis of a cylindrical tube terminated by an open orifice characterized by a nonlinear resistance shows that the reflection function amplitude is reduced as the Strouhal number tends toward unity, but the time delay is not affected. This nonlinear reduction is most important for small-diameter orifices, such as the register hole. Calculations of the saxophone input impedance for second register notes demonstrate that the nonlinear resistance of the register hole reduces the level of the fundamental resonance by 8 dB and lowers its frequency, thereby reducing the degree of harmonicity with higher frequency resonances and enabling the simulated performer to produce a tone in the upper register. Other resonances are not affected. Similar behavior is predicted for multiphonics.

10:05

8MU3. A research program on the "tenora," a Catalan folk woodwind. Ana Barjau (Dept. d'Enginyeria Mecànica, Univ. Politècnica de Catalunya, Diagonal 647, 08028 Barcelona, Spain)

The "tenora" is a Catalan folk woodwind with a conical bore and a double reed. The modern "tenora" evolved from the medieval tenor shawm to fit the needs of more demanding composers and players. This evolution has been mainly artisanal, and as a consequence there is not a standardized design for the instrument. Although much better than old shawms, all tenoras show a non-negligible degree of imperfection in the tuning, stability, and spontaneity of many notes. In order to improve this instrument without impairing its identity, a small group in the Department of Mechanical Engineering of Barcelona started a research program that includes three main parts: experimental measurements (acoustical impedance, impulse response, starting transients, and tuning) to characterize its acoustical identity, theoretical model of the instrument (bore model and reed model) and simulation of it, and redesign algorithm to improve the existing instrument. This research program and its latest results will be presented in this paper. The

attention is focused on the last two points, whose development has led to interesting theoretical work applicable to other instruments.

10:35

8MU4. Acoustics of handbells. Thomas D. Rossing (Dept. of Phys., Northern Illinois Univ., DeKalb, IL 60115), H. John Sathoff (Dept. of Phys., Bradley Univ., Peoria, IL 61625), and Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809)

Although handbells date back at least several centuries B.C., tuned handbells of the type played today were developed in England in the 18th century. One early use of handbells was to provide tower bellringers with a convenient means to practice change ringing. In more recent years, handbell choirs have become popular in schools and churches—some 2000 choirs are reported in the USA alone. In the so-called English tuning of handbells, followed by most handbell makers in England and the United States, the (3,0) mode is tuned to three times the frequency of the (2,0) mode. The fundamental (2,0) mode radiates a rather strong second harmonic partial, however, so that the sound spectrum has prominent partials at the first three harmonics. Modes of vibration of handbells have been studied by means of holographic interferometry, scanning the near-field sound with a microphone, and by modal analysis with impact excitation. Handbells are usually scaled so that the diameter is inversely proportional to the square root of frequency, except in the smallest bells in which it varies inversely with the cube root of frequency. The thickness h is then adjusted so that h/d^2 is nearly proportional to the frequency.

11:00

8MU5. Handbells: "The Belles." Janet Seamons (2648 S. Wood Hollow, Bountiful, UT 84010)

Some of the unique acoustical properties of handbells will be demonstrated with short musical selections.

11:25

8MU6. Acoustics of Caribbean steel pans. Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 61625), Thomas D. Rossing (Dept. of Phys., Northern Illinois Univ., DeKalb, IL 60115), and D. Scott Hampton (Intersonics, Inc., Northbrook, IL 60062)

The foremost musical instrument in its home country of Trinidad, steel pans are becoming very popular in the United States and Europe as well. Steel pans are usually played in ensembles ranging in size from 6 to 8 players to 100 or more. Steel pans are usually fabricated from 55-gal oil drums, and are known by various names, such as lead, tenor, double tenor, guitar, cello, and bass. The sound spectra are rich in harmonic overtones, which appear to have three different physical origins: (1) radiation from higher modes of vibration in a given note area; (2) radiation from nearby notes whose frequencies are harmonically related to the struck note; and (3) nonlinear motion of the note area vibrating at its fundamental frequency. Modes of vibration of both single note areas and the entire drum have been studied by holographic interferometry and by modal analysis with impact excitation.

11:50

8MU7. Steel drums: "Steel Appeal." Darren Duerden, Jennifer Duerden, and Jim Jackson (7205 Station Creek, Midvale, UT 84047)

Some of the unique acoustical properties of steel drums will be demonstrated in the performance of short musical selections.

Session 8NS

Noise: Small Arms Noise Assessment and Mitigation

Paul D. Schomer, Chair
USA—CERL, P.O. Box 4005, Champaign, Illinois 61824-4005

Chair's Introduction—8:30

Invited Papers

8:35

8NS1. Noise protection measures on ranges for small arms using sound attenuating ceilings. Edmund Buchta (Inst. fur Larmschutz, Arnheimer Str. 107, 4000 Dusseldorf, 31, Germany)

In the vicinity of shooting ranges for small arms, people are highly annoyed by impulsive shooting noise. Only the combination of shielding frames and attenuating ceilings leads to noise protection with sufficient efficiency. Such a ceiling should not prevent the natural illumination and may not stop the circulation of air with respect to the removal of exhaust gas of the ammunition. A coffered ceiling, consisting of crossing baffle plates, meets these requirements. Experimental data for ceilings with different size parameters show that excess attenuation in residential areas at distances from 1.0 to 2.0 km of more than 15 dB(A) could be achieved with acceptable efforts.

9:00

8NS2. Theoretical aspects on noise attenuation efficiency of coffered ceilings. K. W. Hirsch and E. Buchta (Inst. fur Larmschutz, Arnheimer Str. 107, 4000 Dusseldorf, 31, Germany)

Coffered ceilings are used as noise abatement measures over shooting ranges for small arms. Usually these ceilings cover the entire ranges using a constant net size between the crossing baffle plates amid a constant plate height. With respect to the muzzle blast from the standard shooting positions and with respect to the sonic boom along the bullet path, size and height could be changed locally without losing attenuation in order to save material or costs, respectively. A "figure of merit" is needed to control a computer optimization process. This figure of merit must predict the relative changes in efficiency with respect to variation of size and height of the baffle plates with sufficient agreements with experimental data. A definition of such a figure of merit and some results of interest will be discussed.

9:25

8NS3. Comparative evaluation of two structures for mitigation of small arms noise. Larry Pater (U.S. Army Construction Eng. Res. Lab., 2902 Newmark Dr., Champaign, IL 61826-9005), Eric R. Sandeen, George Swenson, Jr. (Dept. of Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Kenneth M. Eldred (KEE Engineering, Concord, MA)

An evaluation was made of the noise reduction achieved by two types of noise shielding structures for small arms ranges. One of these was an open front shed with the firing line located within the shed, i.e., a partial enclosure of the firing line. The second structure was a wall barrier located behind the firing line. Theoretical insertion loss was calculated by means of two analytical techniques, one based on the FHWA highway barrier design algorithm and the other a classical diffraction analysis in a spherical coordinate system. Experiments were performed using a shed 7 m high by 6 m deep by 20 m long with a 5.56-mm rifles as noise sources. For guns, which exhibit strong free-field directivity, the experimental results show and the analytical results likewise indicate that directly to the rear the firing shed does not significantly outperform a simple wall.

9:50

8NS4. Penalties for small arms noise assessment. Paul Schomer (U.S. Army Construction Eng. Res. Lab., 2902 Newmark Dr., Champaign, IL 61826-9005)

Proper assessment of the noise created by Army testing and training remains a question that is not fully answered. The most difficult noises to assess are the impulsive noises generated by large weapons, small arms, and helicopters. Currently, general community noise is assessed using the A-frequency weighting and

some form of "energy equivalent" level. In the United States, the day-night average sound level is used. For clearly impulsive sound, adjustments or "penalties" are sometimes added to the formulation to account for the increase in annoyance generated by the impulsive character of the sound. One purpose of the present test was to further define and develop adjustments or "penalties" that can be used to assess military noise *vis-à-vis* normal, urban noises. The study was designed as a paired comparison test where the subjects were presented pairs of sounds and asked, for each pair, which was more annoying, the first or the second sound. The test sounds came from one of three categories of military sources that were: (1) tracked vehicles, (2) small arms fire, or (3) large blasts. The other sound in a pair was one of two control sounds that were: (1) the sound of a wheeled vehicle passing by, or (2) a computer-generated noise. This paper concentrates on small arms noise. The results are for measurements made outdoors—even when the subjects are situated indoors. This is considered to be the proper location since common noise such as traffic is normally assessed on the basis of outdoor levels. The small-arms penalty averages about 8–10 dB across conditions of windows closed or open or subjects outdoors.

Contributed Papers

10:15

8NS5. Noise protection measures on military training facilities of the German Bundeswehr. Gerhard Heiwolt (Germany Federal Ministry of Defense, Ministerialrat, Bundesministerium der Verteidigung, Referatsleiter-U III 2-, Trier Str. 70-72, D-5300 Bonn 1, Germany)

More than a decade ago it was believed that the same noise abatement measures that were established for traffic or industrial noise would also work for shooting noise in the vicinity of rifle ranges. In the early 80's MoD Bonn began to study and to enhance new abatement facilities with respect to the special requirements for shooting noise. The studies apply to constructions, now regarded as "state of the art" so far as Germany is concerned. This report discusses the first constructions to reduce muzzle blast and projectile shock (sonic boom) by enclosing the ranges with sound absorbing ceilings and walls. Furthermore, special measures were developed to reduce sound reflection at structures on or near the ranges. A selected combination of such measures is now successful in protecting a residential area in a certain direction. Some examples of noise reduced shooting ranges in Germany are introduced including some remarks on how to use the structural principles on American-style training facilities in Germany. MoD will continue the support of investigations on noise abatement measures for small arms. There are still 40 military facilities in Germany that need noise abatement measures by law. The technical concept for the German Bundeswehr will be introduced.

10:35

8NS6. Noise reduction using coffered ceiling models. Wayne Bradley (Dept. of Elec. and Comput. Eng., Wave Propagation Lab., Univ. of Illinois, Urbana, IL 61801)

A scale (1 ft = 1 m) model of a German coffered ceiling [E. Buchta, "Noise Control Study of a Light and Air Permeable Coffered Ceiling at an MG-Range in Northeim," Institute for Noise Control, Dusseldorf, Germany (1985)] and three variations were designed, built, and tested at CERL. To measure the efficiency of these ceilings, a scale representation of the spectral range for small arms fire was radiated through the suspended models. The noise reduction measured for these models is compared to theoretical design curves calculated by Cremer [L. Beranek, *Noise and Vibration Control* (Inst. of Noise Control Eng., Washington, DC, 1988)] for attenuation in lined ducts. When the ceilings are considered to be arrays of short lined ducts, Cremer's theory predicts the experimental noise reduction. Attenuation is found to be linearly dependent on the depth of the ceiling. In order to attenuate lower frequencies, the spacing between baffles must be increased. Therefore,

size constraints will determine a lower bound on frequencies effectively attenuated by such ceilings.

10:55

8NS7. Measurement and control of noise from small arms firing facilities. Bennett M. Brooks (United Acoustics Consultants, 27 Hartford Turnpike, Vernon, CT 06066)

Noise due to small arms fire can have a significant impact, both to participants and on the surrounding community, causing considerable annoyance if not controlled. Measurement techniques must allow for the accurate capture of these high-energy impulses, particularly when regulatory forces are involved. Using these test data, control strategies employing progressively more effective means may be developed. In this paper, measured noise data from several small arms firing facilities are presented, including the firing range of a major arms manufacturer. Noise control methods and the results of their application are discussed.

11:15

8NS8. Computer-based gun and blast noise monitoring systems and algorithms. A. M. Snell (Cirrus Res. Ltd., Acoustic House Bridlington Rd., Hunmanby, North York YO14 0PH, England)

This paper describes the use of computer-based noise monitoring systems, the methods of data acquisition, and some new ideas on the analysis of the data for detection of gun noise blasts. The principles and methods are suitable for medium and large area installations. With many modern noise monitoring systems, large amounts of data are acquired for post-analysis. This suggests the use of computers and micro-based instruments to remove the tedium and to actually make the analysis possible. The principle involved in this type of system is to acquire raw data throughout the day (or any other suitable period) from a number of noise monitoring terminals around the site. These data are automatically transferred to the computer for analysis. There is nothing new about this. The difficulties arise in differentiating gun and blast noise generated at the site from, for example, a car door slamming nearby. Elaborate frequency correlating algorithms can be used. These *may* work but at the expense of complexity and, of course, cost. Everyone knows the saying that "The simple methods are always the best." With this in mind, a detection method has been developed using a twin-channel instrument logging short L_{eq} and peak. With one monitor close to the known source and others located around sensitive areas, algorithms have been produced and put into practice, which allow the reliable measurement of small, medium, and large arms around varying areas.

Session 8PA

Physical Acoustics: Finite Amplitude Propagation, Surface Waves

Victor W. Sparrow, Chair

Graduate Program in Acoustics, Pennsylvania State University, University Park, Pennsylvania 16802

Contributed Papers

7:45

8PA1. Quantification of sonic-boom distortion resulting from propagation through the Earth's turbulent boundary layer. Thomas A. Gionfriddo (Graduate Prog. in Acoust., Penn State Univ., 117 Appl. Sci. Bldg., University Park, PA 16802)

Using digitized sonic-boom waveforms from aircraft overflights recorded by the Air Force (Edwards AFB, 1987), a method has been developed to quantify the amount of distortion present in a sonic boom, using a classic N -wave model for comparison. A statistical relation between waveform distortion and the amount of turbulence a sonic boom encounters during propagation is sought. Using the premise that energy conservation applies approximately for a sonic boom both before and after passage through the Earth's turbulent boundary layer, a model N wave is constructed from the distorted boom signature received at the ground. The mean-squared error between the actual and constructed signatures serves as a measure of waveform distortion and is seen to correlate with the distance the wave travels through the turbulence. [Work supported by NASA-LRC.]

8:00

8PA2. A numerical model for the propagation of sonic booms through an inhomogeneous, windy atmosphere. Leick Robinson (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029)

The ZEPHYRUS computer model has been developed to calculate sonic boom propagation through a realistic atmosphere. This model includes the effects of (1) nonlinear distortion (weak shock theory), (2) attenuation and dispersion due to thermoviscous effects and oxygen and nitrogen molecular relaxation, and (3) an inhomogeneous, stratified atmosphere with a stratified wind field. Propagation through reflections and weak (linelike) caustics are included. Analysis of the stability of shocks shows that the characteristic formation distance for stable shocks may be greater than previously believed. In general, stable shocks have not formed when the sonic boom reaches the ground. Also, the rise time has been demonstrated to have a strong dependence on the waveform shape immediately behind the lead shock. Investigations have revealed an interesting phenomenon in which the rise time displays a second peak at the midlevel altitudes, both before and after reflection. This is apparently due to the dispersion from oxygen molecular relaxation when the characteristic relaxation time closely matches the lead shock rise time. [Work supported by NASA and ARL-UT IR&D program.]

8:15

8PA3. Application of a modified NPE model to the prediction of sonic boom waveforms. Andrew A. Piacsek and Allan D. Pierce

(Graduate Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

The nonlinear progressive wave equation (NPE) model, developed by McDonald and Kuperman [J. Acoust. Soc. Am. **81**, 1406-1417 (1987)] has been adapted for atmospheric propagation in order to model the effect of turbulence on the shape of sonic boom profiles. As a test of the efficacy of this adaption, the program was run with a relatively simple scenario conceived in a theoretical analysis by Pierce [J. Acoust. Soc. Am. **44**, 1052 (1968)]. In that paper, spiked and rounded variations of N -wave shock profiles were shown to result from focusing and defocusing, respectively, of an initially rippled wave front propagating in a homogeneous medium. The acoustic pressure versus time profiles generated by the NPE are compared to those predicted by Pierce. Specific improvements and modifications to the NPE are discussed, as well as the intended use of the model to simulate propagation of N waves through an inhomogeneous medium. [Work supported by NASA.]

8:30

8PA4. Parabolic approximations for coherent low-frequency acoustic propagation through atmospheric turbulence. I. W. Schurman (The Johns Hopkins Univ., Appl. Phys. Lab., 7-357, Johns Hopkins Rd., Laurel, MD 20723-6099), W. L. Siegmann, M. J. Jacobson (RPI, Troy, NY 12180), and J. S. Robertson (U.S. Military Acad., West Point, NY 10996)

A parabolic equation (PE) method for the prediction of coherent low-frequency acoustic propagation through small-scale atmospheric turbulence is presented. Frequency constraints on the applicability of stochastic parabolic approximations are avoided by first averaging the stochastic Helmholtz equation and then applying a parabolic approximation to the resulting deterministic equation. Turbulence effects are incorporated by means of spatially varying effective wave numbers. Comparison of exact solutions in the case of infinite-space propagation demonstrates the advantages and limitations of this approach. A uniform asymptotic expression for the effective wave-number profile in the case of isotropic turbulence is used to develop a half-space PE formulation that is valid in the limit of low-frequency, small-scale inhomogeneity. For anisotropic turbulence that is correlated more strongly in range than height, a modified mean-value theorem for the 2-D Helmholtz operator is used to find the effective wave number. Numerical examples demonstrate that excess attenuation due to the imaginary component of the effective wave number is the primary effect of weak turbulence on coherent low-frequency propagation. [Work supported by the U.S. Army Atmospheric Sciences Laboratory through the United States Military Academy and by NASA.]

8PA5. An analysis of low-frequency wind noise in turbulent environments. Scott Morgan and Richard Raspet (Phys. Acoust. Res. Group, Dept. of Phys. and Astron., Univ. of Mississippi, University, MS 38677)

Simultaneous measurement of wind noise and the instantaneous wind speed were performed for bare and screened microphones outdoors. Analysis of these measurements demonstrates that the dominant source of pressure fluctuations at the microphone outdoors is the intrinsic turbulence in the flow. This is in contrast to the results of measurements performed in low turbulence environments by Hosier and Donovan and by Strasberg. For low turbulence conditions the fluctuating wake of the screen is the dominant noise source. This finding has important implications for windscreen design for outdoor measurements since the principles described by Hosier and Donovan apply only to low turbulence conditions.

9:00

8PA6. Application of similarity theory for calculating the acoustic refractive index structure parameter. John M. Noble (U.S. Army Atmospheric Sci. Lab., ATTN: SLCAS-AM-A, White Sands Missile Range, NM 88002)

The acoustic refractive index structure parameter is important for determining the strength of atmospheric turbulence. The current methods for calculating the acoustic refractive index structure parameter is to calculate the temperature and wind structure parameter from tedious hot-wire anemometer measurements. Also most methods only provide a measurement of the turbulence strength at one height. An alternative method to calculate the refractive index structure parameter is to calculate the temperature and wind structure parameters using similarity theory. The similarity model requires easily obtained meteorological data at one height along with an estimate of the roughness length. The model uses similarity theory to calculate the temperature and wind structure parameters with height. This provides the capability to calculate the refractive index structure parameter with height. A comparison with turbulence data [Johnson *et al.*, *J. Acoust. Soc. Am.* **81**, 638-646 (1987)] will show how well the model works. This type of model predicts the turbulence strength with height through the surface layer for different stability states of the atmosphere.

9:15

8PA7. A fast-field algorithm for sound propagation in layered media using a two-dimensional FFT with iterative refinement. Michael J. White (U.S. Army Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826-9005) and Y. L. Li (Univ. of Illinois at Urbana-Champaign, 1406 W. Green St., Urbana, IL 61801)

A technique is presented for evaluating the field from an assembly of monopole sources embedded in a medium of homogeneous planar layers. Each layer is characterized by a sound speed, density, absorption coefficient, and thickness, and the first and last layers use an impedance

condition. The main differences between this and previous implementations of the fast-field method are formulation for multiple sources, use of the two-dimensional Fourier transform, lack of any far-field restriction, and the ability for iterative refinement of the inverse transform for the pressure. Iterative refinement allows for convergence testing of the field solution without discarding previous results, making it easy to choose an optimum sampling of the spatial inverse Fourier transform. Some examples will be shown to demonstrate the ability of the new technique to solve for the near field and far field of directional sources above ground in a refracting atmosphere. Potential applications include ground impedance investigations via acoustic propagation, and prediction of the field radiated by a helicopter on its warm up pad.

9:30

8PA8. On indirect acoustic impedance computations involving fixed free-space attenuation or fixed boundary loss factors. Adeboyejo A. Oni (Lab. for Advanced Industrial Concepts, Clarence W. Mitchell School of Eng., Morgan State Univ., Baltimore, MD 21239)

A class of methods for speedily estimating the acoustic impedance of surfaces is presented. The methods utilize a lattice search algorithm to estimate the impedance that will validate analytical expressions for ground attenuation of sound, using experimental measurements of continuous-wave sound attenuation changes between two above-ground sound source locations with respect to a single microphone close to the ground. In the two cases considered, opportunities appear to exist for a considerable reduction in computation complexity: in the first case, the arrangement allows spherical spreading components of the excess attenuation terms to essentially cancel out. In the second case, the arrangement allows the boundary loss factor to remain constant while the source sound location changes. Variations in meteorological parameters between the two sound source locations are assumed negligible. These proposed analytical and experimental arrangements display the potential to provide a means of obtaining speedy and efficient estimates of ground impedance for various surfaces. [Work supported by the U.S. Army Research Office.]

9:45

8PA9. Wave scattering from a rough ground in an inhomogeneous medium. Y. L. Li and S. J. Franke (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Rm. 60G, Everitt Lab., 1406 W. Green St., Urbana, IL 61801)

An exact analytical expression of the Green's function is obtained for the case of a time-harmonic *line source* embedded in a medium with the linear sound-speed profile. This is the first time an analytical two-dimensional Green's function has been found for wave propagation in an inhomogeneous medium. Substituting the exact analytical expression of the Green's function into the Helmholtz-Kirchhoff integral equation and using the method of moments, the scattered fields from a ground with barriers or troughs in such a medium are computed. The effects of scattering in the region beyond the caustic surface are also studied. [Work supported by U.S. Army Construction Res. Lab.]

10:00-10:15

Break

10:15

8PA10. The propagation of plane N waves through a statistically isotropic turbulent field as a singular perturbation problem. Bart Lipkens (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029 and Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX 78712-1063)

Reported here is a theoretical investigation to explain experimental observations of the effect of turbulence on N -wave propagation [B. Lipkens and D. T. Blackstock, *J. Acoust. Soc. Am.* **90**, 2307(A) (1991)]. An adaptation of von Kármán's spectral model for incompressible, isotropic turbulence is used to generate a statistical realization of a turbulent field. The 3-D, random, isotropic velocity field consists of a collection of discrete Fourier velocity modes [M. Karweit *et al.*, *J. Acoust. Soc. Am.* **89**, 52-62 (1991)]. A lossless wave equation is derived that incorporates the effect of the turbulence field [see, for example, K. Plotkin, Ph.D. thesis, Cornell University (January 1971)]. The turbulence is assumed to be frozen during passage of an initially plane N wave. A regular perturbation expansion in the turbulence Mach number is used to solve the problem. At second order the problem becomes singular, and coordinate straining is applied to remove the singularity. The first-order perturbation makes the N wave ragged. The second-order perturbation appears to introduce changes in the phase speed. [Work supported by NASA.]

10:30

8PA11. Propagation of pulsed finite amplitude sound beams in a liquid with strong absorption. Michalakos A. Averkiou, Yang-Sub Lee, and Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78712-1063)

Measurements of intense acoustic pulses generated by directive sources in a liquid with strong absorption are compared with theoretical predictions based on a time domain numerical solution of the KZK nonlinear parabolic wave equation [Lee and Hamilton, in *Ultrasonics International 91 Conference Proceedings* (Butterworth-Heinemann, Oxford, 1991), pp. 177-180]. The computer program is useful for describing the propagation of tone bursts with various amplitude and frequency modulations. Experiments were performed in glycerin with narrow-band pulses having center frequencies of several megahertz. The pulses experience self-demodulation that leads to far-field waveforms characterized by the low-frequency envelope, as predicted by the asymptotic theory of Berklay [*J. Sound Vib.* **2**, 435-461 (1965)]. Good agreement between theory and experiment was obtained for amplitude-modulated tone bursts, not only at far-field axial positions as achieved first by Moffett *et al.* [*J. Acoust. Soc. Am.* **49**, 339-343 (1971)], but also off axis, throughout the transition region and into the near field. The effects of frequency modulation are also discussed. [Work supported by the ONR and the David and Lucile Packard Foundation.]

10:45

8PA12. Molecular relaxation effects on the thickness of weak shock waves in seawater. Jongmin Kang and Adnan Akay (Acoustics, Vibrations and Noise Control Lab., Dept. of Mech. Eng. and Inst. for Manufacturing Res., Wayne State Univ., Detroit, MI 48202)

Many works have been done for predicting peak pressure and the decay time constant of underwater explosion waves; however, little efforts had been made for determining a detailed structure of the leading edges. Marsh *et al.* [*J. Acoust. Soc. Am.* **38**, 326-338 (1966)] investigated the effects of the finite amplitude and $MgSO_4$ relaxation on the explosion waveforms in seawater, and showed the anomalous increase of the rise time from their measurements (from 5 to 12 μs). Recently, a molecular relaxation model for determining a rise phase of weak shocks propagating through the relaxing atmosphere was developed by Kang and Pierce, where O_2 and N_2 are dominant relaxation processes. The theory is applied to the rise phase of weak shocks in seawater, in which,

equivalently, two relaxation processes are important: $MgSO_4$ and $B(OH)_3$ [Fisher and Simmons, *J. Acoust. Soc. Am.* **62**, 558-564 (1977)]. The molecular relaxation in seawater depends on temperature, ambient pressure, salinity, and pH. Numerical simulations show the dependency of shock thickness on peak pressure and local properties of seawater such as temperature, ambient pressure, and salinity, and also clarify the individual role of each relaxation process and the classical absorption.

11:00

8PA13. Observation of a new surface mode on fluid-saturated porous solids. Peter B. Nagy (Dept. Welding Eng., Ohio State Univ., Columbus, OH 43210)

Almost 10 yr ago, Feng and Johnson predicted the presence of a new surface mode on a fluid/fluid-saturated porous solid interface [*J. Acoust. Soc. Am.* **74**, 906 (1983)]. The most interesting feature of this new mode is that, as a true guided mode, its velocity is lower than any of the bulk velocities in the surrounding media, including the slow compressional wave. Experimental observation of this mode is rendered very difficult by the requirement that the pores at the surface be closed. It was found that, due to surface tension, practically closed-pore boundary conditions can prevail on the "free" surface of most fluid-saturated solids. Strong capillary forces at the boundary between the nonwetting (air) and the wetting (water or alcohol) fluids extend an ideally thin but very stiff membrane over the otherwise open-surface pores. The surface mode was directly excited by a vertically polarized shear transducer mounted at the edge of the sample and detected by a heterodyne laser interferometer. Surface wave velocity and attenuation measurements were made on both synthetic and natural porous solids in the frequency range between 50 and 300 kHz. The experimental results provide clear evidence of the new "slow" surface mode predicted by Feng and Johnson and show the feasibility of using this mode for characterization of permeable formations.

11:15

8PA14. Acoustic surface waves propagating horizontally above a seasonal snow cover. Donald G. Albert (U.S. Army Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755-1290)

Experimental measurements of acoustic impulses propagating horizontally above a seasonal snow cover reveal the presence of acoustic surface waves that were not observed under grass or frozen ground conditions. The measurements were obtained using blank pistol shots fired 1 m above the snow as the source of acoustic impulses; the resultant acoustic waveforms were measured by microphones at heights of from 0.1 to 4.75 m after 60 m of horizontal propagation along the surface. The snow cover was 0.20 m thick with an average density of 200 $kg\ m^{-3}$ and crystal sizes of 0.5 to 1 mm. The peak pressures measured at all of the microphones were markedly reduced compared to pressures measured when a snow cover was not present because of the well-known absorptive effect of the snow. In addition, waveforms near the surface displayed a strong, low-frequency "tail" following the impulsive arrival from the shot. This tail was found to decay exponentially with height z above the surface [$\sim e^{-\alpha z}$], a diagnostic feature of surface waves, with a measured attenuation coefficient $\alpha \approx 0.5\ m^{-1}$. Waveforms calculated using Attenborough's four-parameter, layered model of ground impedance are shown to agree with the observed waveforms when an assumed surface effective flow resistivity of 20 $kN\ s\ m^{-4}$ is used. [Work supported by the Directorate of Research and Development, U.S. Army Corps of Engineers, Project 4A161102AT24.]

11:30

8PA15. Scholte wave generation. Jean Duclos and Alain Tinel (Lab. d'Acoust. Ultrason. et d'Electron., L.A.U.E., U.R.A. C.N.R.S. 1373, Université Le Havre, Place Robert Schuman, 76610 Le Havre, France)

The Scholte–Stoneley interface wave is interesting in an acoustic field because it does not decay during its propagation. Different techniques have been used to generate this liquid/elastic solid interface wave. Some of these techniques are based on the conversion of an incident wave, for example a longitudinal wave in the case of the alcohol wedge method. A Scholte wave can be generated directly using an interdigital transducer (I.D.T.) deposited on a piezoelectric substrate, or an amorphous substrate covered with a thin piezoelectric film. The study of the reversibility of the Scholte wave diffraction phenomenon by a dihedral [A. Tinel and J. Duclos, *Acoust. Lett.* **15**, 30–35] leads to the presentation of another technique to generate a Scholte wave. It has been verified that the interaction of an incident bulk wave with the dihedral in a particular direction results in a Scholte wave at the interface. This method is efficient and easy to use.

11:45

8PA16. Experimental study of Scholte wave diffraction by a dihedral.
Alain Tinel and Jean Duclos (Lab. d'Acoust. Ultrason. et d'Electron.,

L.A.U.E., U.R.A. C.N.R.S. 1373, Université Le Havre, Place Robert Schuman, 76610 Le Havre, France)

The theoretical study of waves capable of propagating at a liquid/elastic solid interface may lead to many solutions. Some of these solutions correspond to interface waves that are characterized by their direction of propagation parallel to the interface and by their possible reemission in the fluid medium. After recalling the main characteristics of the Scholte–Stoneley and the Rayleigh waves, the authors propose an experimental study of the diffraction of the Scholte wave by a dihedral edge. When the Scholte wave meets the dihedral perpendicularly, two phenomena occur. First, a phenomenon of pure diffraction characterized by a bulk wave mainly propagating in the interface direction. This is essentially due to the component of the Scholte wave in the liquid. Second, some conversions occur giving a reflected Scholte wave, a transmitted one, and two leaky Rayleigh waves. It is possible to prove reemission directions of the generalized Rayleigh waves in accordance with the solid nature and the dihedral angle.

FRIDAY MORNING, 15 MAY 1992

SALON F, 8:25 TO 11:15 A.M.

Session 8PP

Psychological and Physiological Acoustics: Masking

Neal F. Viemeister, Chair

Department of Psychology, University of Minnesota, 75 East River Road, Minneapolis, Minnesota 55455

Chair's Introduction—8:25

Invited Paper

8:30

8PP1. Contemporary issues in masking hailing from Zwicker's work. Søren Buus (Commun. and Digital Signal Process. Ctr., 409 DA, Northeastern Univ., Boston, MA 02115)

Recent work on masking has focused on cases that appear inconsistent with the power-spectrum model of masking. This model holds that simultaneously masked thresholds are determined by the signal-to-masker ratios within frequency-selective channels. More than a decade ago, Zwicker noted that an amplitude-modulated broadband masker is much less effective than a steady-state masker with the same energy. This effect, known as comodulation masking release (CMR), might be mediated by detection of envelope correlation or by listening during brief periods of low masker intensity (in the valleys). Recent experiments show that CMR is obtained for brief tones presented in the valleys of the masker, but not at the peaks. This finding favors the "listening in the valleys" explanation. Zwicker also was among the pioneers in the exploration of "overshoot" of masking. Recent experiments show that the overshoot depends on the onset of energy outside the critical band centered on the signal and not on energy within it. Finally, Zwicker had a long-standing interest in how detection thresholds depend on signal bandwidth. Recent experiments show that the energy at detection threshold is nearly independent of bandwidth for brief signals, but increases with bandwidth for long-duration signals. This finding indicates that the decision rule used for detection depends on duration. All three phenomena can be modeled if the weights applied to the frequency-selective channels of the power-spectrum model are assumed to vary over time because they are driven by the stimulus. [Work supported by NIDCD.]

9:00

8PP2. Temporal effects with single- and multiple-band maskers: Effects of level. Sid P. Bacon, Gail A. Takahashi, and Michelle L. Hicks (Dept. of Speech and Hear. Sci., Arizona State Univ., Tempe, AZ 85287-0102)

The temporal effect with broadband maskers was dubbed "overshoot" by Zwicker in 1965. Recent experiments suggest that frequency regions quite remote from the signal frequency are important for this effect. The purpose of the present study was to determine which frequency region(s) might be responsible for the observation [S. P. Bacon, *J. Acoust. Soc. Am.* **88**, 698-702 (1990)] that overshoot is nonmonotonic with masker level. The signal was a 10-ms, 4.0-kHz sinusoid presented 1 or 195 ms after the onset of the 400-ms masker. The spectrum level of the masker ranged from -5 to 55 dB. Four masker configurations were employed: lower band (LB, 2.5 to 3.5 kHz), middle band (MB, 3.5 to 4.5 kHz), upper band (UB, 4.5 to 5.5 kHz), and all band (AB, 2.5 to 5.5 kHz). There was a significant temporal effect for all but the MB; it was monotonic with level for the UB, but nonmonotonic for the LB and AB, suggesting that the LB contributes most to the nonmonotonic effect in the AB condition. Preliminary analyses suggest that at low levels (where the temporal effect exists in the AB condition) the masking by the individual bands is combining in a much more nonlinear fashion at the 1-ms delay. [Work supported by NIDCD.]

9:15

8PP3. Intensity discrimination in backward masking. Christopher J. Plack and Neal F. Viemeister (Dept. of Psychol., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455)

It has been demonstrated that the Weber fraction for a brief tone burst is elevated when the tone is preceded by an intense narrow-band noise [F-G. Zeng *et al.*, *Hear. Res.* **55**, 223-230 (1991)]. This elevation is greatest when the tone burst is at medium sound levels (around 40-70 dB). The experiment of Zeng *et al.* was repeated except that the presentation order of the tone burst and the noise was reversed. Intensity discrimination was measured for a 1-kHz tone burst with a steady-state duration of 25 ms and 2-ms onset and offset ramps. The tone burst was presented in quiet or 100 ms before a 200-Hz-wide narrow-band noise centered on 1 kHz with a spectrum level of 70 dB and a duration of 100 ms. Not only was the "midlevel hump" evident in the backward masking conditions, but the effect was larger than in the equivalent forward masking case. This result will be discussed with regard to possible explanations for the existence of the midlevel hump. [Work supported by NIDCD Contract No. DC00683.]

10:15

8PP6. Across-frequency processing of multiple modulation patterns. J. H. Grose and J. W. Hall, III (Div. Otolaryngol./Head & Neck Surgery, Univ. North Carolina at Chapel Hill, Chapel Hill, NC 27599-7070)

9:30

8PP4. Effects of ambient noise masking on earphone thresholds. Dennis L. Williams, Thomas A. Frank, and Marie K. Schmitt (Dept. of Commun. Disord., 110 Moore Bldg., Penn State Univ., University Park, PA 16802)

Monaural thresholds were measured for 24 normally hearing subjects using TDH-49P, Audiocup, and ER-3A earphones from 500 to 6000 Hz in quiet and in background noise having maximum permissible ambient noise levels (MPANLs) specified by OSHA (1983) and ANSI (S3.1-1991). The attenuation for each earphone system was obtained and the actual threshold shift was compared to the predicted threshold shift. The mean threshold shifts were negligible (<2 dB) in the ANSI MPANLs regardless of earphone and in OSHA MPANLs for the ER-3A but ranged from 2 to 7.5 dB for the Audiocup and from 6 to 12.5 dB for the TDH-49P in the OSHA MPANLs. The results of this study suggest that OSHA MPANLs are too high to allow for testing down to 0-dB HL using these earphones. The high noise-reducing properties of the ER-3A would make it the earphone of choice if testing is done in OSHA MPANLs. Testing down to 0 dB HL can be done in the newly specified ANSI MPANLs regardless of earphone type.

9:45

8PP5. An evaluation of the role of envelope-based cues for the detection of a tone added to noise. Virginia M. Richards (Dept. of Psychol., Univ. of Pennsylvania, 3815 Walnut St., Philadelphia, PA 19104) and Robert D. Nekrich (Dept. of Bioengineering, Univ. of Pennsylvania, Philadelphia, PA 19104)

Using 2IFC paradigm, d' scores were obtained for two conditions derived from the tone-in-noise task. In one condition, observers detected a tone added to a band of noise, but the waveforms presented in the two intervals were of equal energy. Noise center frequencies and signal frequencies of 600, 1800, and 5400 Hz and noise bandwidths of 40, 120, and 360 Hz were used. In the second condition, the envelopes of the noise-alone and tone-plus-noise waveforms were extracted, and the envelopes were used to modulate sinusoids of 600, 1800, and 5400 Hz. The observers' ability to discriminate between modulation patterns was measured. For narrow bands of noise, d' values in the two conditions were approximately equal, indicating that for level-equalized waveforms the detectability of a tone added to noise could be accounted for by differences in the envelope patterns. For bandwidths that were larger than or equal to 20% of the center frequency, d' scores obtained using amplitude modulated tones were superior to those obtained using the level-equalized stimuli, an advantage that may reflect differences in the power spectra of the modulated tones. [Supported by the NIH.]

10:00-10:15

Break

In a study linking comodulation masking release (CMR) and auditory grouping, it was demonstrated that the auditory system is capable of performing a CMR using one modulation pattern while a second modulation pattern is simultaneously present [J. W. Hall and J. H. Grose, *J. Acoust. Soc. Am.* **88**, 119-125 (1990)]. This study proceeds to examine whether the auditory system can perform a CMR on a complex

signal that is jointly masked by separate modulation patterns; i.e., can the auditory system perform a CMR using each of two concurrent modulation patterns? The signal was a two-tone complex (618 and 1400 Hz). The composite backgrounds were composed of narrow bands of noise whose center frequencies were harmonics of either 103 or 175 Hz. Various noiseband configurations were constructed to result in modulation patterns that were either adjacent, interspersed, or completely uncorrelated across bands. Results to date indicate that thresholds are markedly lower in conditions where the noisebands form two adjacent regions of comodulation relative to conditions where the bands are uncorrelated. CMR is reduced when the modulation patterns are interspersed. [Work supported by the Deafness Research Foundation.]

10:30

8PP7. CMR in complex masking backgrounds for normal and cochlear hearing-impaired listeners. J. W. Hall, III and J. H. Grose (Div. Otolaryngol./Head & Neck Surgery, Univ. North Carolina at Chapel Hill, Chapel Hill, NC 27599-7070)

Comodulation masking release (CMR) was measured in masking backgrounds composed of multicomponent comodulated noise bands, with a variable number of bands (deviant bands) present that had a modulation pattern different from those of the comodulated set. Normal-hearing listeners showed results that were in agreement with previous findings [J. W. Hall and J. H. Grose, *J. Acoust. Soc. Am.* **88**, 119–125 (1990)]: CMR was largest when no deviant bands were present; CMR was very small when only a few deviant bands were present; CMR recovered to some extent as more deviant bands were added. These results are interpreted in terms of frequency selectivity, auditory object formation by amplitude envelope coherence, and the ability of a listener to process a signal in the presence of competing background sounds. The data of cochlear hearing-impaired listeners showed considerable intersubject variability. Some of the hearing-impaired subjects had results that were very similar to those of the normal-hearing subjects (relatively large CMRs for conditions where there were no deviant bands, or many deviant bands). Other cochlear hearing-impaired listeners showed relatively small CMRs in general, but most of these listeners still showed an overall effect of improvement with increased number of deviant bands. Results will be discussed in terms of peripheral and central contributions to CMR. [Research supported by NIH NIDCD.]

10:45

8PP8. Masking period patterns in normal-hearing and hearing-impaired listeners. Anna C. Schroder and David A. Nelson (Dept. of Otolaryngol., Univ. of Minnesota, 2630 University Ave. SE, Minneapolis, MN 55414)

Masking period patterns (MPPs), developed by Zwicker [*J. Acoust. Soc. Am.* **59**, 166–175 (1976)] as a psychophysical correlate of the physiological period histogram, were obtained from four normal-hearing and four cochlear hearing-impaired subjects. The just audible level of a brief, relatively high-frequency, test tone was measured as a function of its temporal position within the period of a 40-Hz masker. Zwicker's findings were replicated in normal listeners. At low masker levels a single maximum (Max 1) was observed between 0 and 90 deg of the masker period. At higher masker levels a second maximum (Max 2) developed between 180 and 270 deg. Zwicker's model [*Biol. Cybern.* **23**, 49–60 (1976)] suggests that Max 1 reflects level-independent suppression effects and Max 2 reflects some combination of level-dependent excitation and suppression effects, which is consistent with physiological biasing data. For impaired listeners, in regions of hearing loss, the degree of modulation in the MPP was reduced relative to that seen in normal-hearing listeners. The pattern of modulation was usually altered as well. These results are discussed in the context of outer hair cell damage at the place in the cochlea corresponding to the test tone frequency. [Supported by NIDCD.]

11:00

8PP9. Auditory filter shapes in normal-hearing, noise-masked, and elderly subjects. Mitchell S. Sommers and Larry E. Humes (Speech Res. Lab. and Depts. of Psychol. and Speech and Hear. Sci., Indiana Univ., Bloomington, IN 47405)

Previous studies of frequency selectivity have suggested a strong positive correlation between age and the width of auditory filters [Patterson *et al.*, *J. Acoust. Soc. Am.* **72**, 1788–1803 (1982)]. However, given that absolute thresholds are generally higher in older listeners, it is unclear whether the broader filter shapes are a consequence of aging *per se* or are associated with changes in absolute sensitivity. To dissociate the effects of hearing loss and increased age on changes in frequency selectivity, this study measured auditory filter shapes at 2 kHz in (1) normal-hearing young subjects; (2) elderly (over age 65) subjects with normal 2-kHz thresholds; (3) young subjects with 2-kHz thresholds elevated either 20 or 40 dB by a narrow-band masker; and (4) elderly subjects with varying degrees of hearing loss at 2 kHz. Rounded exponential filter shapes were derived from the data using the method described by Patterson [*J. Acoust. Soc. Am.* **59**, 640–654 (1976)]. Equivalent rectangular bandwidths (ERBs) of auditory filters were not significantly different in young and elderly subjects with normal 2-kHz hearing. Furthermore, filter widths for young subjects with 20- and 40-dB simulated hearing losses overlapped with those obtained from elderly subjects with corresponding degrees of actual hearing loss. These results suggest that the reduced frequency selectivity reported for older listeners can be attributed, primarily, to hearing loss rather than increased age. [Work supported by NIH.]

Session 8SP**Speech Communication and Education in Acoustics: Education in Speech Communication
(Lecture and Poster Session)**

Sigfrid D. Soli, Chair

*House Ear Institute, 2100 West Third Street, Los Angeles, California 90057***Chair's Introduction—8:30*****Invited Papers*****8:35**

8SP1. Educational issues in speech communication. Emily A. Tobey (Dept. of Commun. Disord., LSU Med. Ctr., New Orleans, LA 70112), Winifred Strange (Univ. of South Florida, Tampa, FL 33620), Maureen Stone (The Johns Hopkins Univ., Baltimore, MD 21218), Ken Stevens (MIT, Boston, MA 02139), and Peter Ladefoged (Univ. of California at Los Angeles, Los Angeles, CA 90024)

The Speech Communication Technical Committee has formed a subcommittee to investigate educational issues related to the instruction of speech acoustics. The Education Subcommittee is initially focusing their efforts on core issues that cut across the different disciplines represented in Speech Communication. The purpose of this report is to share the findings of a recent questionnaire sent to the Speech Communication and Psychological and Physiological Acoustics membership. Data will be reported on the scope of acoustic training, the common acoustic topics taught across disciplines, the types of software used to teach speech acoustics, the type of textbooks used, the type of handouts used to augment lectures, the type of video/audio materials used as demonstrations, and the type of laboratory experiences available to students. In addition, this presentation will review the types of laboratory equipment used to train students in basic and speech acoustics.

8:55

8SP2. The perils of teaching speech acoustics. Emily A. Tobey (Dept. of Commun. Disord., LSU Med. Ctr., New Orleans, LA 70112), Winifred Strange (Univ. of South Florida, Tampa, FL 33620), Ken Stevens (MIT, Boston, MA 02139), Peter Ladefoged (Univ. of California at Los Angeles, Los Angeles, CA 90024), and Michael Dorman (Arizona State Univ., Tempe, AZ 85287)

The Speech Technical Committee of the Acoustical Society is composed of individuals from a variety of disciplines including linguistics, psychology, engineering, speech science, and communication disorders. The diversity of disciplines represented in speech communication results in studentbodies with a wide range of background knowledge in basic and speech acoustics. The purpose of this panel discussion is to stimulate a dialogue regarding current teaching practices, problem areas unique to particular disciplines, problem areas common across disciplines, and future directions for improving the instruction of basic and speech acoustics.

Panel Discussion**PANEL MODERATOR: Winifred Strange**

**PANEL MEMBERS: Emily Tobey
Michael Dorman
Peter Ladefoged
Kenneth Stevens**

9:35–9:50**Break**

All posters will be on display and all authors will be at their posters from 9:50 to 11:50 a.m.

8SP3. Development of synthesis-analysis tools for a computer-based course in acoustic phonetics. Robert Berkovitz, J. M. Pickett, and David Williams (Sensimetrics Corp., 64 Sidney St., Cambridge, MA 02139)

Decreasing prices of personal computers and signal-processing boards permit the widespread use of computer-based instruction in acoustic phonetics. Software already developed for the Sensimetrics *SpeechStation* provides simplified speech synthesis and convenient spectral analysis displays. Synthesis exercises in which the student controls a vocal tract diagram are being developed to teach the principles of relations between articulatory configurations and speech sounds. *SpeechStation* analysis facilities will provide correlated displays of spectra of synthetic or natural utterances. Initial plans are to develop a complete introductory course of 16 lessons. Course material for advanced students would serve to train in research methods and teach advanced topics. A trial lesson will be demonstrated. [Work supported by NIDOCOD.]

8SP4. Survey for computer-based course in acoustic phonetics. J. M. Pickett (Windy Hill Lab., Surry, ME 04684) and Robert Berkovitz (Sensimetrics Corp., 64 Sidney St., Cambridge, MA 02139)

Instructors of courses in speech communication listed in the 1990 JASA summary of acoustics courses were surveyed to develop information about numbers and types of students, course names, current text use, available computers, and rating-scale opinions on the potential value of computer-based courses in acoustic phonetics. Responses were overwhelmingly positive. Surprisingly large numbers of courses and students were reported. Equipment was indicated as available for computer-based instruction. Detailed results will be presented on the distribution of courses over academic disciplines and levels of texts in use, to characterize the needs for course development. [Work supported by NIDOCOD.]

8SP5. Project Oracle: Software for training in speech science. J. Anthony Seikel and David G. Drumright (Univ. Prog. in Commun. Disord., Washington State Univ. at Spokane, W. 601 First Ave., Spokane, WA 99204-0399)

Project Oracle was instituted to develop public domain software and laboratory activities that would give students in the speech and hearing sciences the opportunity to explore speech and nonspeech acoustics in a nonthreatening environment. Waveform editing software and lab activities were developed for speech science and phonetics courses and in-

roduced into the curriculum. The presentation will provide examples of lab activities, hands-on demonstration of the waveform editing, FFT, and phonetics transcription training software, as well as results of software implementation. [Work supported by Dept. of Education, FIPSE.]

8SP6. Waveform editing and spectrographic software. Stephen T. Neely (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131), Susan Nittrouer (Univ. of Nebraska at Omaha), and Edward J. Carney (Boys Town Natl. Res. Hospital)

Programs for interactive editing and spectrographic display of digitized speech waveform files have been developed for use on IBM-PC compatible computers. The waveform editing program (WavEd) reads and writes waveform files of several types, including ILS sampled data. WavEd allows the user to perform the following operations: (1) record and play waveforms; (2) place labels at various points, and save these labels; (3) copy, delete, and insert waveform segments; and (4) do acoustic analyses, such as computation of durations, formant frequencies and bandwidths, and linear prediction coefficients. The spectrographic display program (SPECTO) produces spectrograms from ILS sampled data files using filters of user-specified bandwidths. Parameters that determine spectrogram appearance (such as darkness, contrast, and frequency and time range) are user controlled. Spectrograms can be sent to the PC monitor or to dot-matrix or laser printers. Both programs are designed to be easy to install and to use on any PC, with either a commercial digital-to-analog converter or the PC-internal speaker. Both programs are being distributed as unsupported software and nonprofit redistribution is permitted. [Work supported by NIH.]

8SP7. Teaching acoustics to phoneticians. Peter Ladefoged (Phonetics Lab., Dept. of Linguistics, UCLA, Los Angeles, CA 90024-1543)

A hypercard stack has been written that provides an interactive approach to the basic concepts of acoustics. It is assumed that the students will know little or no mathematics. Graphical displays are provided of wave motion and the superimposition of waves. Students can both see and hear the effect of adding waves of their choice. Concepts such as the Nyquist frequency, vibration of air in a tube and in the vocal tract, and the parameters required for synthesizing speech are demonstrated. A section of the stack is concerned with the properties of damped exponential waves. The notions of the addition of sine and cosine waves and the correlation between two waves are demonstrated, as a basis for understanding Fourier analysis. The stack is copyrighted, but is available for nonprofit use at a nominal charge from the author.

Session 8UW

Underwater Acoustics: Geoacoustics of Laterally Inhomogeneous Sediments

Robert D. Stoll, Chair

Lamont-Doherty Geological Observatory, Columbia University, Palisades, New York 10964

Chair's Introduction—7:55

Invited Papers

8:00

8UW1. Seafloor shear wave velocity variability. LeRoy M. Dorman (Marine Phys. Lab., Scripps Inst. Oceanogr., UCSD, La Jolla, CA 92093-0215)

In gross classification, sediments of the deep seafloor may be of terrigenous or marine origin. Those of marine origin are classified as oozes (generally calcareous) or clays (generally siliceous) depending largely on whether their CaCO₃ content is greater or less than 30%. The carbonate content is, in turn, controlled both by the biological productivity of the surface waters and the circulation patterns of the deep waters of the oceans. Older waters have more dissolved CO₂ and attack carbonates more aggressively. There are now a few deep-sea *in-situ* shear velocity measurements in siliceous and calcareous sediments. The siliceous sediments show velocities that are the lowest, 20–30 m/s at the surface, while calcareous sediments have slightly higher velocities. Both of these have velocities lower than those found in shallow silts and sands. Measurements of the shallow shear velocities show high variability, approaching 50%, over a few km distance in the siliceous sediment (Dorman *et al.*, 1991). These measurements are made using interface waves from seafloor explosions observed using ocean bottom seismographs. These may be correlated with the channeling of sediments but the mechanism is far from clear. [Work supported by ONR.]

8:25

8UW2. High-resolution estimates of spatially variable compressional and shear velocities within marine sediments. Joseph F. Gettrust and Mary M. Rowe (Naval Res. Lab., Code 7430, Stennis Space Center, MS 39529)

Multichannel seismic reflection data are used to obtain estimates of compressional and shear velocities within marine sediments. Estimates of compressional velocities are made by interactively computing stacking velocities that transform the reflection hyperbola for reflection horizons to a straight, horizontal line. These stacking velocities are used to compute interval velocities that estimate the mean compressional velocity between two reflection horizons. The compressional velocity within the water column is known through direct source-to-hydrophone path travel times; typically, these direct path measurements are within 1% of the interval velocities computed from the seafloor reflection. This suggests that the estimates of the compressional velocity within the sediments are within a few percent of the "true" interval velocity, well below the 10% to 15% spatial variability over distances as short as 20 m that are resolved with these data. Shear velocity estimates computed using spectrum versus offset analysis show that the spatial variability of shear velocity over similar distances often is greater than 30%. [Work supported by the Office of Naval Technology and ONR.]

8:50

8UW3. Spatial variability of surficial shallow-water sediment geoacoustic properties. Michael D. Richardson (Naval Res. Lab., Stennis Space Center, MS 39529)

In a previous publication [Richardson, in *Ocean Seismo-Acoustics* (Plenum, New York, 1985), pp. 527–536] the author reviewed relationships between physical and biological processes and the resultant variability of values of sediment geoacoustic (compressional wave velocity and attenuation) and physical (grain size and porosity) properties. In this presentation the subject is revisited, using values of compressional and shear wave velocity measured *in situ* instead of the laboratory measurements considered in the previous paper.

8UW4. Effects of lateral inhomogeneity on the geoaoustic properties of sediment. Robert D. Stoll and George M. Bryan (Lamont-Doherty Geological Observatory of Columbia Univ., Palisades, NY 10964)

Dispersion analysis of interface waves (Scholte/Stoneley/Rayleigh) has been used extensively to determine geoaoustic models of near-bottom ocean sediments. In many cases a single trace has been analyzed to derive group velocity dispersion using a multiple filter technique. The dispersion curve obtained in this manner reflects the average influence of the sediment acoustic properties over the entire path from source to receiver. However, when a linear array of receivers is used to obtain data simultaneously at a number of different ranges, it is possible to derive both group and phase velocity dispersion curves that correspond to propagation over a short portion (typically 5 m) of the total path at many different ranges. With this kind of data, the range-dependent variation of geoaoustic properties in the sediment may be assessed. Results are presented based on data obtained with a 24-receiver array and an explosive source, both deployed on the seafloor. Analysis of data from a number of different locations, mainly in shallow water ranging in depth from a few meters to 250 m, suggests that there is a considerable amount of lateral inhomogeneity even over very short ranges. [Work supported by ONR, Code 11250A.]

9:40-10:00

Break

10:00

8UW5. Intrinsic attenuation and scattering in laterally inhomogeneous sediments. R. A. Stephen (Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

Intrinsic attenuation represents energy lost from the acoustic field through internal friction. Scattering represents energy lost from a particular coherent path either to another coherent path (as in compressional to shear wave conversion) or to incoherent energy (e.g., angle spread, time spread, or coda). Nothing can be done with energy lost to intrinsic attenuation. However, in the case of multiple coherent paths processing could be carried out to optimize the total energy on all paths. In the case of incoherent scattering there is an increase in the "noise" floor due to the source (source-generated noise) that will yield an upper bound on the signal-to-noise ratio independent of the true ambient noise. All three contributions to "effective" attenuation will have different frequency dependence and the relative importance of each mechanism will depend on the environment. In order to thoroughly understand the observed geoaoustic field in marine sediments it is necessary to distinguish between these effects. A propagation modeling code has been developed, based on the method of finite differences, which incorporates intrinsic attenuation. The same code can treat scattering from volume heterogeneities and surface roughness for structures with scale lengths comparable to acoustic wavelengths. All propagation effects including shear waves, interface waves, two-way propagation, and multiple interactions between scatterers are considered. The first problem that will be addressed with this code is the effect of intrinsic attenuation on the coda (time spread) caused by scattering from volume heterogeneities in sediments. [Work supported by ONR.]

10:25

8UW6. The measurement of modal evolution in a range-dependent, shallow-water waveguide. George V. Frisk, Kazuhiko Ohta, Subramaniam D. Rajan, and James A. Doust (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543)

A technique is described for the measurement of normal mode eigenvalues as a function of range in a laterally inhomogeneous, shallow-water waveguide. It is based on the notion, originating in adiabatic mode theory, that the local modes adapt to the local environment. The trajectories of the eigenvalues in range are therefore influenced by lateral variations in the geoaoustic parameters of the seabed as well as bathymetric changes and fluctuations in the acoustic properties of the water column. Knowledge of the modal evolution in a waveguide plays a critical role in characterizing sound propagation and its relationship to the environment in shallow water. The method consists of beamforming narrow-band pressure field data acquired on a horizontal array using a short sliding aperture in range and examining the trajectories of the modal peaks in the beamformed output. The modal resolution can be significantly enhanced by mode filtering with a vertical array of receivers prior to the beamforming operation. Applications to synthetic and experimental data are discussed. [Work supported by ONR.]

10:50

8UW7. Lateral variability of geoaoustic properties of the seafloor. Dennis Lindwall and Joseph Gettrust (Naval Res. Lab., Code 7430, Stennis Space Center, MS 39529)

Using deep towed, multichannel seismic reflection data, the spectral response of the seafloor sediment as a function of offset or grazing angle along several kilometers of old Atlantic seafloor near the Blake outer ridge has been measured and variations of as much as 15 to 20 dB in bottom loss over distances of less than 50 m have been found. A video will be shown of this spectrum versus offset (SVO) data that explicitly shows the lateral variations as the DTAGS instrument is towed along just above the seafloor. A few regions of anomalously high shear wave velocities also will be pointed out. These data demonstrate that a seemingly smooth seafloor can be a rough, frequency-dependent scattering surface. [This work is supported by the Office of Naval Technology.]

11:05

8UW8. A measurement of the grain bulk modulus of sands. Joseph C. Molis and Nicholas P. Chotiros (Advanced Sonar Div., Appl. Res. Labs., P.O. Box 8029, Austin, TX 78713-8029)

The grain bulk moduli of several different types of sand from various locations were measured. The method involves compressing a known volume of sand, held in an aluminum bowl, to about 13.78 MPa (2000 psi), using castor oil as the liquid medium, and measuring the volume of released liquid as a function of pressure. The apparatus consists of a pressure chamber, pressure sensor, release valve, and a graduated tube to measure the volume of released fluid. Pressure is decreased and the escaping volume of castor oil is measured, in about constant intervals of 0.344 MPa (50 psi). The change in pressure with respect to released volume and the measured volume of the sand sample are used to determine the grain bulk modulus. Sand samples from various locations have been analyzed. Previous models of acoustic propagation in sandy ocean sediments used an assumption that the grain bulk modulus of sand grains are equal to that of quartz. These results show that sand is significantly more compressible. [Work supported by Naval Oceanographic and Atmospheric Research Laboratories.]

11:20

8UW9. Shear wave transducer for ocean sediment applications. Alan K. Walden and Thomas R. Howarth (Naval Res. Lab., Underwater Sound Reference Detachment, P.O. Box 568337, Orlando, FL 32856-8337)

A low-frequency shear wave transducer has been developed for ocean geoaoustic applications. The transducer generates and receives torsional shear waves for increased energy propagation. The design is a lightweight structure with a hollow cylinder sandwiched between two radiating disks to form a spool-shaped geometry. The interior and exterior are encapsulated with a soft polymer filler. Piezoelectric ceramics are mounted on the cylinder and configured to excite the "1-5" mode of the piezoceramic during operation. The radiating disks are designed to "pull" the encapsulant medium such that they introduce a torsional shear wave through the encapsulant and into the sediment medium. Low-frequency operation is ensured through the proper selection of the structure's dimensions and materials. The design suggests that performance will be improved as the hydrostatic pressure is increased.

11:35

8UW10. A two-way parabolic equation method for elastic media. Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

A two-way parabolic equation (PE) method for acoustic media [J. Acoust. Soc. Am. **91**, 1357-1368 (1992)] is generalized to elastic media. A range-dependent ocean is approximated by a sequence of range-independent regions. The one-way elastic PE is used to propagate the transmitted and reflected fields through each of the regions. A scattering method based on the one-way elastic PE is used to compute transmitted and reflected fields at the vertical boundaries between regions. The scattering method is implemented using an efficient iteration formula for the reflected field that is analogous to the iteration formula for the two-way acoustic PE method. The two-way elastic PE method is useful not only for studying backscattering but also for achieving improved accuracy for the outgoing signal.

11:50

8UW11. Conversion of seismic waves in inhomogeneous and anisotropic sediments. H. Schmidt and H. Fan (MIT, Cambridge, MA 02139)

Traditionally theoretical and numerical modeling in ocean bottom seismo-acoustics has assumed the sediments to be horizontally stratified, allowing for integral transform or wave-number integration methods to be used, as for example the SAFARI code [H. Schmidt and F. B. Jensen, J. Acoust. Soc. Am. **77**, 813-825 (1985)]. Although this assumption is often valid, there is experimental evidence that such simplified environmental models are not always correctly representing the seismo-acoustic behavior. Thus the seasonal nature of the sedimentation process introduces a fine layering that is usually ignored. However, such a sediment will not be isotropic, even at low frequencies, but exhibit transversely isotropic behavior. Here use is made of a modified version of SAFARI to demonstrate the effect of this anisotropy on the seismo-acoustics. Another, even more significant effect is introduced by lateral inhomogeneities giving rise to scattering as well as seismic wave type conversion. For example, horizontally polarized Love waves are often observed in experimental data, a feature that is not predicted by the simple two-dimensional theories. Here, a 3-D extension of an elastic boundary element code [P. Gerstoft and H. Schmidt, J. Acoust. Soc. Am. **89**, 1629-1642 (1991)] is used to demonstrate how the excitation of the Love waves can be explained by the presence of such discrete, lateral inhomogeneities in the stratification.

12:05

8UW12. Spectral parabolic equation solutions of three-dimensional problems involving horizontal coupling and backscattering. G. J. Orris, Michael D. Collins, and W. A. Kuperman (Naval Res. Lab., Washington, DC 20375)

The parabolic equation (PE) method and separation of variables are applied to solve problems in which the ocean-acoustic parameters are independent of one of the horizontal Cartesian coordinates. This class of problems is representative of backscattering from ridgelike seamounts, horizontal coupling out of the radial direction, and other types of reverberation. Since the PE method efficiently handles nearly arbitrary variations in depth and in the other horizontal direction, it is possible and practical to solve complicated three-dimensional problems with this approach. The two-way PE is applied to backscattering problems. Solutions are compared with three-dimensional PE solutions for problems involving horizontal coupling. Problems involving elastic ocean bottoms are also considered.

8UW13. Data processing for high-resolution 3-D subbottom imaging system "kite." Tokuo Yamamoto and Howoong Shon (Univ. of Miami, Geo-Acoust. Lab., RSMAS, 4600 Rickenbacker Cswy., Miami, FL 33149)

A high-resolution 3-D subbottom imaging system called "Kite" has been successfully tested in shallow ocean near Miami Beach in July 1991. Kite is a 24-channel hydrophone array towed laterally, i.e., the axis is perpendicular to the direction of the ship motion. An omnidirectional piezoelectric source located at the center of the hydrophone array is excited with a half-wave at 4 kHz every second while towed at 1 m/s. Each hydrophone signal is recorded at 16 kHz in 16 bits using an IBM compatible computer and an A/D converter. The 3-D image processing is done by using a software for conventional multi-channel seismic reflection data. First, the 3-D image of compressional wave velocity is constructed. The density and shear wave velocity images are constructed assuming the normal consolidation stress condition [Yamamoto *et al.*, *Geophys. J. Int.* **98**, 173–182 (1989)]. The results show that Kite has a great advantage in that it produces in high-resolution 3-D subbottom images while its data can be handled in a conventional way. [Work supported by ONR.]

8UW14. Nonlinear inversion of sediment parameters in a range-dependent region of the Gulf of Mexico. Michael D. Collins, W. A. Kuperman (Naval Res. Lab., Washington, DC 20375), and Henrik Schmidt (MIT, Cambridge, MA 02139)

A set of acoustic data was taken in the Gulf of Mexico and used to invert sediment properties by Lynch *et al.* [*J. Acoust. Soc. Am.* **89**, 648–665 (1991)]. Using a linear optimization method and a range-independent propagation model, they obtained sediment parameters that result in fairly good agreement between the measured and theoretical transmission loss (TL) curves. Since the data are for a long synthetic-aperture horizontal array, it is ideal for range-dependent inversion. Using simulated annealing for nonlinear optimization and the parabolic equation method for range-dependent propagation, sediment parameters have been obtained that result in excellent agreement between the measured and theoretical TL curves. The range dependence of the depth of the interface between the upper and lower sediment layers was determined to be a particularly important parameter.

Session 9AB

Animal Bioacoustics: Nonhuman Primate Acoustical Communications

Marc D. Hauser, Chair

Animal Communication Laboratory, University of California, Davis, California 95616-8761

Chair's Introduction—12:55

Invited Papers

1:00

9AB1. The role of experience in nonhuman primate vocal development. Michael J. Owren (Dept. of Psychol., Univ. of Colorado at Denver, Campus Box 173, P.O. Box 173364, Denver, CO 80217-3364)

In contrast to the extensive evidence of plasticity in avian and human vocal development, studies in nonhuman primates present a conflicting picture of the normal course of vocal ontogeny. Evidence from squirrel monkeys (*Saimiri sciureus*) indicates that calls emerge fully formed at an early point and that little or no modification of calling behavior occurs. Investigations involving Old World monkeys have produced evidence of modifiability in both signal form and use. However, none of these studies have directly shown the necessity of particular experiences for the emergence of species-typical vocal behavior. Two recent experiments have sought direct evidence through cross-fostering of infant rhesus (*Macaca mulatta*) and Japanese (*M. fuscata*) macaques between species, but have provided inconsistent results. Overall, the evidence continues to indicate that experience plays a comparatively limited role in nonhuman primate vocal ontogeny. [Work supported by NIH.]

1:20

9AB2. Acoustic correlates of social contexts and inferred internal states in howling monkeys (*Alouatta palliata*). James M. Whitehead (Inst. for Advanced Study of the Commun. Processes, Univ. of Florida, Gainesville, FL 32611)

Students of animal communication since Darwin have hypothesized that emotions find expression in animal utterances. This assertion is being tested in an examination of tape recordings of a well-studied primate, the mantled howling monkey, and of the natural contexts in which calls occur. Previous work has shown that, by means of sound playbacks, monkeys can be induced to approach the sound of another calling, hence possibly rival, male. Since approach to males is normally followed by prolonged vocal battles and sometimes by combat, it is hypothesized that acoustic correlates of stress or affective states such as fear or arousal will be evident in such features as lowered fundamental frequency, alteration of vocalization rate, instability of voice, etc. Although the study is incomplete, examination of the spectrum of roars, measured by FFT and LPC, has failed to reveal consistent differences. Possible reasons for the lack of acoustic differences include technical limitations, constraints on mechanisms of loud call production, and selection pressure against revealing intentions. [Supported by the H. F. Guggenheim Foundation.]

1:40

9AB3. A comparative approach to the nonhuman primate vocal tract: Implications for sound production. Miguel A. Schoen Ybarra (Dept. of Anatomy, School of Medicine, Univ. of Puerto Rico, San Juan, PR 00936-5067)

This talk aims at elucidating the sound-producing capabilities inherent to the nonhuman primate vocal tract morphology. Focus is on tract features which, on the bases of their respective form, structure, and location, can be regarded as (1) capable of contributing to generate a voiced/unvoiced source function, or (2) to transiently modify the transfer function of the tract. Relevant qualitative data, and, whenever accessible, also quantitative, were obtained from current references, and in the laboratory of the speaker where morphology-to-performance relationships are being studied in the vocal tract of the rhesus macaque and some New World monkeys. Comparative considerations suggest that the vocal tract of nonhuman primates can produce a variety of sounds and sound contrasts common to all nonhuman primates, but, also, that some of the taxa can make certain sounds and sound contrasts which others cannot.

9AB4. Neuroethological approaches to the study of vocal production in nonhuman primates. John D. Newman (Lab., of Comparative Ethology, Natl. Inst. of Child Health and Human Development, NIH, Poolesville, MD 20837-0289)

The vocalizations of nonhuman primates are potentially important probes for studying the neural and neurochemical substrates of behavior. These studies have focused on one category of vocalization, the isolation call, due to its relatively stereotyped acoustic structure and ease of elicitation and attribution in captive animals. Experimental brain lesions reveal an important role for the thalamic tegmentum and anterior cingulate gyrus in mediating normal acoustic structure and motivation to vocalize, respectively. The amygdala plays a role in determining the development of normal levels of emotional expression, as shown in the effects of neonatal ablations of this limbic structure and its cortical afferents on vocal behavior during development. Neurochemical control of the isolation call arises from several disparate endogenous neurochemical systems, including opiate and monoaminergic pathways.

9AB5. The role of articulation in the production of nonhuman primate vocalizations. Marc D. Hauser (Animal Commun. Lab., Univ. of California, Davis, CA 95616-8761)

Observational and experimental data on rhesus monkeys are presented that demonstrate that changes in articulation, especially movements of the lips and mandible, have a direct influence on the resonance properties of the call, independently of changes in the fundamental frequency. These data suggest that the source-filter theory of sound production is also applicable to nonhuman primates. In the second part of the presentation, results are discussed from experiments that examined the ability of rhesus monkeys to compensate for novel vocal tract perturbations. In particular, having demonstrated that the lips play a role in sound production under natural conditions, acoustic analyses of the vocalizations produced by animals whose lips were temporarily paralyzed through the administration of lidocaine were conducted. Results showed that for some vocalizations, lidocaine failed to cause detectable changes in acoustic morphology whereas for other vocalizations, which appeared to involve more fine-grained manipulations of lip configuration, there were noticeable acoustic consequences. Together, these results suggest that nonhuman primates have significantly greater articulatory freedom than heretofore predicted.

9AB6. Are primate vocalizations adapted to the local habitat? Charles H. Brown, Rafael Gomez (Dept. of Psychol., Univ. of South Alabama, Mobile, AL 36688), and Peter M. Waser (Dept. Biol. Sci., Purdue Univ., West Lafayette, IN 47907)

Twenty-one representative vocalizations of two species of rain forest monkeys [blue monkeys (*Cercopithecus mitis*) and grey-cheeked mangabeys (*Cercocebus albigena*)], and two species of savanna monkeys [vervet monkeys (*C. aethiops*) and yellow baboons (*Papio cynocephalus*)] were broadcast and re-recorded in both the "appropriate" and "inappropriate" habitat in Kenya. The re-recorded signals were digitized and distortion analyses were conducted with signal processing routines on a Cray XMP-24 supercomputer. Rain forest monkey calls were distorted less in the "appropriate" rain forest habitat than in the "inappropriate" savanna habitat, but the converse did not hold for savanna monkey calls. Savanna monkey calls were distorted the same amount in both habitats (time domain analysis), or were distorted slightly less by the rain forest habitat (frequency domain analysis). The results suggest that the rain forest is more favorable than the savanna for high-fidelity sound propagation, and, in addition, rain forest monkey calls appear better "designed" than savanna monkey calls to reduce the probability of distortion by the acoustics of the local habitat. [Work supported by NIDCD.]

9AB7. Comparisons of monkey and human sensitivity to speech, nonspeech, and monkey speech sounds. Joan M. Sinnott (Comparative Hear. Lab., Dept. of Psychol., Univ. of South Alabama, Mobile, AL 36688)

Psychoacoustic studies are reviewed comparing monkey and human differential sensitivity to a wide variety of simple and complex auditory signals. (1) Nonspeech: for the basic psychoacoustic capacities of pure tone frequency (ΔF), intensity (ΔI), and temporal (ΔT) discrimination, monkeys discriminate $+\Delta I$ nearly as well as humans, but are moderately less sensitive to ΔT , and markedly less sensitive to ΔF and $-\Delta I$. (2) Speech: Monkeys show varying degrees of sensitivity to complex human speech sounds. Monkeys are as sensitive as humans to English liquid /ra-la/ contrasts, moderately less sensitive to English VOT /ba-pa/, place of articulation /ba-da/, stop-glide /ba-wa/, and various vowel contrasts, but markedly less sensitive to prevoiced Spanish VOT contrasts. (3) Monkey-speech: Monkeys are moderately less sensitive

than humans to temporal variation in pitch peak position along a macaque coo continuum. Taken together, all these data indicate a good correspondence between monkey sensitivity to ΔI and ΔT in simple and complex sounds. However, this is not the case for ΔF , since monkeys appear more similar to humans in discriminating spectral variation in complex sounds than in simple pure tones. [Work supported by NIDCD.]

3:20-3:40

Panel Discussion

PANEL MODERATOR: Phil Lieberman

FRIDAY AFTERNOON, 15 MAY 1992

SALON C, 1:00 TO 3:00 P.M.

Session 9ID

Interdisciplinary: Late Papers

Irvin G. Bassett, Chair

Department of Physics, 296 ESC, Brigham Young University, Provo, Utah 84602

Contributed Papers

1:00

9ID1. Dynamic behavior of a piezoelectric dry femur. M. Cengiz Dökmeci (Istanbul Tech. Univ., P. K. 9, Taksim, 80191 Istanbul, Turkey)

This paper is concerned with a mathematical model for the dynamic behavior of a piezoelectric dry femur. The dry femur is taken as a cylindrical bar of approximately circular cross section. The mechanical displacements and the electric potential of the dry femur are expanded in a series of Jacobi polynomials in terms of the polar coordinates of the cross section. These expansions are inserted into a generalized variational principle [M. C. Dökmeci, IEEE Trans. Ultrason. Ferroelec. Freq. Control 35(6), 775-787 (1988)] so as to consistently deduce a hierarchy of one-dimensional equations of the dry femur from the three-dimensional equations of piezoelectricity. These macroscopic equations, which are expressed in both variational and differential forms, accommodate all the extensional, radial, flexural, and torsional motions as well as coupled motions of femur. The uniqueness of solutions in the governing equations of femur is investigated and the initial-mixed boundary conditions are enumerated which are sufficient for the uniqueness. Cases involving special motions are considered, some conclusions regarding the results obtained are drawn, and future needs of research are indicated. [Work supported in part by the U.S. Army through its European Research Office.]

1:15

9ID2. Quantitative aspects of acoustic emission in theory and practice. Mark A. Friesel (Battelle, Pacific Northwest Lab., P.O. Box 999, Richland, WA 99352)

Acoustic emissions are known for their sensitivity to conditions of

the sources, propagating media, and detection instrumentation. The concurrent ease of use of this technology for nondestructive evaluation has helped foster the notion that acoustic emission techniques are incurably qualitative by nature. Recent theoretical developments and use of modern signal processing and analysis techniques, however, show that acoustic emissions are rich sources of quantitative information about active microscopic damage and other processes. After discussion of the current state of acoustic emission theory, the characteristics of certain data types will be examined. The results obtained by applying various signal processing methods to acoustic emission data are presented.

1:30

9ID3. Acoustic scattering from fluid turbulence. Louis Goodman, Diane Szargowicz (Naval Undersea Warfare Ctr., Newport, RI 02841), John Oeschger (Univ. of Rhode Island, Kingston, RI 02881), and Michelle O'Donnell (Americom Corp., Middletown, RI 02840)

There has been considerable interest in the oceanographic community in whether it is possible to use high-frequency acoustic echo sounders to measure ocean microstructure. A laboratory program, the ocean acoustics turbulence study (OATS), has been developed to examine the nature of such scattering and to compare results with that of a model recently proposed by Goodman (1990). The fundamental theory underlying the model is Bragg scattering, which results in the scattered pressure field proportional to the Fourier transform of the temperature field. A small heating element is used to generate a buoyant turbulent plume. Such a relationship could allow the prediction of the three-dimensional turbulent wave-number spectrum provided a sufficiently

wide dynamic range in Bragg wave-number space were measurable. By performing scattering experiments over a very wide range of angles ($\vartheta = 5^\circ$ to $\vartheta = 160^\circ$) and frequencies (250 kHz to 1.4 MHz) it is possible to infer such a composite spectrum. Preliminary results indicate a turbulent wave-number spectrum of classical form ($-11/3$ power law) in agreement with that expected for a fully developed turbulent field.

1:45

9ID4. Acoustical estimation of bubble concentration in seawater. Dong Ding (Shanghai Acoust. Lab., Shanghai, 200032, People's Republic of China)

The acoustic backscattering characteristics of seawater were studied by the method of local acoustic measurements within a small volume to infer the bubble concentration in seawater. It was found that the scattering intensities were increased with wind speed, especially to the forming of white caps on the sea surface. In some cases the intensities were related to the sea state of earlier days. Under low wind speeds, the peak of the scattering spectra shifted toward high frequencies and its spectra became narrower. In high winds the spectra reversed its tendencies. The spectra in deep water at high winds had a similar shape as in shallower depth at light winds. The measured data can be fitted by the theoretical spectra constructed on the principles of energy superposition under the assumption of Poisson distribution of bubble number in different sizes per unit of volume. The dependence of bubble concentration on depth can be obtained by the fittings. The concentrations exponentially decrease with the depth. The decaying constant is 8 (1/m) and is independent of wind speed. A definition of effective thickness of the bubble layer near the sea surface was given by a critical density of 10^{-3} (1/cm³), $\Delta R = 1 \mu\text{m}$. The inferred thickness is dependent on the wind speed.

2:00

9ID5. Some aspects of wavelet transform in biosonar signal processing and detection. Zhen-Biao Lin (Department of Electrical Engineering, Purdue University at Indianapolis, 723 W. Michigan St., Indianapolis, IN 46202)

Wavelet transform (WT) provides multiresolution at different frequencies. The concept of WT is related to the constant Q filter bank in modeling of cochlea, and is therefore suitable for the study of auditory signal processing in bats. The results from FM bat signal analysis show that FM bat signals consist of a number of scaled basic prototype (wavelets): from a low-frequency component with long duration to a high-frequency component with short duration or large bandwidth. For example, the signals in the search phase, which tunes in time-frequency resolution, allow the bat to extract diverse target information at different frequencies, global information (motion) at low frequency and distance information at high frequency. The bat's signals in terminal phase consist of 2nd and 3rd harmonics in a fraction of duration of 0.2–0.5 ms. Therefore an insect's precise location may be obtained at these harmonic frequencies. In a case where the bat-emitted signal is selected as the basic prototype of wavelet, the WT becomes a version of the wide-band ambiguity function. Two dozen signals emitted by an FM bat, *Pipistrellus*, during natural hunting of insects were recorded and analyzed with WT and Wigner distribution (WD). Two-dimensional delay scalograms and two-dimensional time spectrums in different hunting phases are given. The strong link between WD and WT is used to explore the strategies of echolocation signals adopted to diverse tasks. The Doppler tolerant waveform is a special selection of wavelet basic prototype, such that all "wavelets" have the same version as that of the prototype.

2:15

9ID6. Atmospheric sound propagation using a three-dimensional parabolic equation. Yves Delrieux (Office National d'Etudes et de

Recherches Aérospatiales, B. P. 72, 92322 Châtillon Cedex, France)

A numerical method for solving the three-dimensional parabolic wave equation (3DPE) is presented to investigate long-range sound propagation in the atmosphere near the ground. An implicit finite-difference scheme based on the alternating directions method is formulated. The pressure field is calculated in cylindrical coordinates for all the azimuth ranges from 0° to 360° , so that only a periodicity condition is required. A hard reflecting ground is considered and a homogeneous boundary condition at the top of the mesh is applied. The validity of the 3DPE is shown by comparison with measurements carried out in an anechoic chamber. The diffraction of sound behind a thin barrier of finite length is studied. A good agreement with calculations is obtained for the sound pressure level behind the barrier. In particular, the term involving azimuth derivatives is taken into account in the 3DPE and enables one to properly calculate the diffraction from both side edges of the screen. Moreover, calculations show that the presence of wind may modify the position of interferences created by the screen. [Work supported by Direction des Recherches Etudes et Techniques.]

2:30

9ID7. Acoustic property measurement of ultrasonic delay line. Yinguan Wang (Institute of Acoustics, Tongji University, 1239 Siping Rd., Shanghai, 200092, People's Republic of China)

The acoustic properties of F05 ultrasonic delay-line glass produced by the Shanghai Xin-Fu Glass Factory are measured with an instrument built by this institute. The instrument can automatically and continuously measure a small variation of ultrasonic velocity by a pulse superposition (PS) method with a solution of 2×10^{-7} ; precisely measure absolute ultrasonic velocity by a pulse echo overlap (PEO) method with a precision of 1×10^{-4} ; and automatically and continuously measure ultrasonic attenuation by a gated echoes comparison (GEC) method with a resolution of 0.1 dB. Three parts in one instrument not only can possess the advantage of convenience, but also can overcome the shortage of determining the absolute value by the PS method. The absolute velocity of longitudinal wave in the F05 delay line is 4140 (m/s) (all at 20°C), temperature coefficient of travel time is $-3.3 \times 10^{-5}/^\circ\text{C}$, attenuation coefficient is $0.12 + 0.02$ (dB/cm), and temperature coefficient of attenuation is $2.5 \times 10^{-3}/^\circ\text{C}$. The absolute velocity of a transverse wave in the F05 delay line is 2524 (m/s), the temperature coefficient of travel time is $-4.5 \times 10^{-6}/^\circ\text{C}$, attenuation coefficient is $2.4 + 0.04$ (dB/cm), the temperature coefficient of attenuation is $6 \times 10^{-3}/^\circ\text{C}$.

2:45

9ID8. An ultrasonic spike pulse emission circuit. Jingsheng Wei (Grain Storage Res. Inst., Commercial Ministry, No. 97 Hua Pai Fang 610031, Chengdu City, Sichuan Province, People's Republic of China)

For many ultrasonic testing instruments, the piezoelectric transducers are working under a pulse situation. To improve the resolution of these instruments, the designs of the transducers must be considered. Recently, some ultrasonic instruments such as blood flow instruments, a Doppler system that uses ultrasonic pulsed random signals, and spectral analysis in food ultrasonography all emit and receive wideband spike pulses, so that they need piezoelectric transducers and emission circuits with a wider frequency band and higher efficiency. In this paper use is made of a Redwood equivalent circuit and a Laplace transform to analyze the impulse response of the emitting and receiving networks for the thickness mode piezoelectric transducers. The theoretical analysis shows that the sensitivity of the acoustic system is proportional to the square of the electromechanical coupling coefficient and inverse to the flight time of the ultrasonic wave within the transducer. An ultrasonic spike pulse emission circuit for thickness mode piezoelectric transducers was installed on the basis of theoretical analysis. The experimental results are in good agreement with the theoretical results.

Session 9PA

Physical Acoustics: Scattering

W. Pat Arnott, Chair
Desert Research Institute, Reno, Nevada 89506

Contributed Papers

1:00

9PA1. Investigation of scattering with acoustic holography. Ming-Te Cheng, Anna L. Pate (Dept. of Aerosp. Eng. and Eng. Mech., Iowa State Univ., Ames, IA 50011), and Dayle Groutage (DTRC Det. Puget Sound, Bremerton, WA 98314-5212)

The objective of this paper is to investigate sound scattering of a rigid sphere using the acoustic holography method. A piston source, which is vibrating sinusoidally, illuminates a sphere. The superimposed incident and scattered fields are measured in the near field of the sphere using acoustic holography. These two fields are decomposed in the wave-vector domain and, therefore, the scattered field is extracted. Numerical simulations are performed and the effect of various parameters is investigated. Specifically, the distance between two holography planes, 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 cast-iron sphere as a scatterer. The experiments demonstrate the feasibility of the field separation technique based on two-plane acoustic holography. [Work supported by David Taylor Research Center.]

1:15

9PA2. The determination of the relative strengths of resonances excited for plane waves scattered from elastic spheroids. C. E. Dean, M. F. Werby, and Elmer White (NRL, Codes 221 and 223, Stennis Space Center, MS 39529)

Calculations are presented for scattering from spheroids composed of six materials for aspect ratios of 3 and 6. The difference in resonance locations is noted as a function of material properties and the differences can be explained in terms of pseudo-Rayleigh type resonances described extensively in the literature for spheroids. There is also a marked difference between the amplitudes of the resonances. It is possible to predict the relative amplitudes of the resonances as a function of material properties based also on the Rayleigh wave interpretation of these resonances. A theory is developed based on the fact that Rayleigh type resonances can only be excited as a function of the appropriate grazing angular region. This region is determined as a function of the material property and geometry. The theory accounts for the numerical predictions nicely. Numerous results are presented.

1:30

9PA3. The manifestation of certain classes of resonances as standing waves on elastic spheroidal objects. M. F. Werby, C. E. Dean, and Elmer White (NRL, Code 221 and 223, Stennis Space Center, MS 39529)

In an earlier work it has been shown that certain classes of reso-

nances excited on elastic spherical solids correspond to standing wave patterns on the object surface. This observation was in fact implicit in the circumferential nature of such resonances. The demonstration was made possible by subtracting the rigid background and plotting the bistatic angular distributions in the asymptotic limit resulting in a standing wave pattern. For the spheroidal case it is more difficult to demonstrate this effect in the asymptotic limit since typically the asymptotic limit implies spherical symmetry that is not adhered to for spheroids. It is possible to demonstrate that this is also observed for resonances on spheroids by choosing a spheroidal surface and plotting the results once the rigid background is subtracted. It is also possible to observe the results by projecting the asymptotic results on to a spheroidal basis. Results are presented for several examples.

1:45

9PA4. Signatures from pulse signals scattered from elongated elastic solids. Elmer White, M. F. Werby, and C. E. Dean (NRL, Code 223 and 221, Stennis Space Center, MS 39529)

Time domain signals can manifest characteristic signals when the mid-pulse frequency is in some resonance region. This work attempts to determine unique signals characteristic of pulse scattering at frequencies associated with resonances excited from elongated targets. In an earlier work beat patterns and damped sinusoidal patterns (as a function of time) were associated with single or specific clusters of resonances. Here, damped sinusoidal patterns are associated with specific resonances in which it is shown how to extract specific resonance widths that are shown to be associated with three classes of resonances excited from elastic spheroids. This includes resonances excited broadside, end-on, and at oblique incident angles. Also determined are some beat patterns from some types of resonance clustering.

2:00

9PA5. A comparative study of the performance of time-frequency representations for the analysis of echo returns. Roger H. Hackman, Theagenis J. Abatzoglou, Hal Arnold, and J. Reagan (Lockheed Palo Alto Res. Labs., Palo Alto, CA 94304-1191)

In a scattering experiment, the contributions (pulses) associated with different dynamical scattering processes generally arrive at the field point at different times. The structure of these individual pulses is related to the characteristics of the underlying scattering dynamics (the "dynamical" bandwidth of the coupling, underlying wave dispersion, etc.). Thus the time-frequency structure of the echo return is intimately connected not only with the frequency bandwidth and shape of the incident pulse, but also with the target response. Elements of the above have been discussed by previous authors [e.g., N. Yen *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **84**, S185 (1988)]; in this paper, a comparative study is presented of the analysis capabilities of several time-frequency anal-

yses tools, including the Wigner distribution, Choi-Williams distribution, the Gabor transform, and continuous wavelet algorithms. The study is based on echo returns that have been synthesized from a numerical T -matrix solution for a finite steel cylinder. The target echoes in this study may generally be characterized as multicomponent, time-varying signals with a time-bandwidth product that is not large.

2:15

9PA6. Reflection of focused sound beams from curved surfaces. Michalakis A. Averkiou and Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78712-1063)

The reflection of focused sound beams from surfaces with spherical curvature is investigated theoretically and experimentally. Theoretical predictions for the incident and reflected beams are based on the parabolic wave equation. A circular source with a uniform amplitude and quadratic phase distribution is assumed. Solutions for the reflected beam are derived for both pulsed and continuous sources. The experiments were performed in water with a 3.5-MHz source that has a nominal radius of 2.5 cm and focal length of 15 cm. Accurate measurements of the incident beam, particularly very near the source, were used to characterize the effective radius and focal length. Reflection from both convex and concave surfaces was investigated. The targets were made of nickel with radii of curvature that vary from 5 cm up to infinity (planar targets). Measurements of the reflected beam were obtained with a needle hydrophone that passed through a small hole in the center of the source. Agreement between theory and experiment is excellent, and the results suggest novel ways to measure surface curvature. [Work sup-

ported by the David and Lucile Packard Foundation, ONR, and NSF.]

2:30

9PA7. Imaged travel time surface of the transverse cusp caustic produced by reflection from a curved surface and the shape of reflected transients. Carl K. Frederickson (Phys. Acoust. Res. Group, Univ. of Mississippi, University, MS 38677) and Philip L. Marston (Washington State Univ., Pullman, WA 99164)

As a curved wavefront propagates in a uniform medium it may form caustics. Catastrophe theory can be used to classify the structurally stable caustics, where the initial local shape of the wave front determines the classification. These caustics have an associated travel time surface that describes the merging and disappearance of rays as the caustic is crossed. The travel time surface is defined by the travel time of signals propagating along rays from the wave front to the observer. Experiments were performed to image an acoustical transverse cusp diffraction catastrophe along with its associated travel time surface. The finite difference in arrival times of the acoustical signals allows the resolution of the travel time surface. Reflection of a spherical source signal from a curved surface in water was used to produce an initial wave front with the general local shape that produces a transverse cusp. Calculations of the expected travel time surface compared well to images cut through the experimental surface. Acoustical signals traveling along rays that touch the caustic are shown to have the Hilbert transform shape of the incident signal. Those signals that do not touch the caustic have the same general shape as the incident signal. [Work supported by ONR.]

2:45-3:00

Break

3:00

9PA8. Axisymmetric wave propagation in fluid-loaded cylindrical shells: Theory versus experiment. Thomas J. Plona, Bikash K. Sinha, Sergio Kostek, and Shu-Kong Chang (Schlumberger-Doll Res., Old Quarry Rd., Ridgefield, CT 06877)

This paper discusses both theoretical and experimental aspects of axisymmetric wave propagation along fluid-loaded cylindrical shells (excluding torsional modes). For a steel cylindrical shell with a fixed ratio of inner to outer radius, four different fluid configurations are considered: water inside and outside the steel shell; air inside and outside; water inside and air outside; and air inside and water outside. Calculations of the transient pressure response for the case of an axisymmetric ring source and a point receiver are made as a function of source-receiver separation. Experimentally, a PZT ring source and ring receiver are placed around a steel cylindrical shell with an outer radius of 9.53 mm and an inner radius of 7.94 mm. Waveforms are recorded for multiple source-receiver hydrophone spacings in the frequency band 50-240 kHz. Using a Prony's method, the complex wave number as a function of frequency for each of the modes in the system is derived from both the theoretical and experimental waveforms. These axisymmetric modes are grouped into three categories: steel modes, non-cutoff fluid modes, and cutoff fluid modes. Excellent agreement is obtained between theory and experiment.

3:15

9PA9. Acoustic scattering from cylindrical shells bounded by hemispherical endcaps. Dominique Décultot, Fernand Léon, Florence Lecroq, and Gérard Maze (Lab. d'Acoust. Ultrason. et d'Electron., L.A.U.E., U.R.A. C.N.R.S. 1373, Université du Havre, Place Robert Schuman, 76610 Le Havre, France)

The acoustic scattering from infinite cylinders or spheres has shown the great influence of surface wave propagation. Two types of surface waves are distinguished: the Rayleigh or Whispering Gallery waves and the Scholte-Stoneley waves. In this study, experimental results obtained on the finite cylindrical shells bounded by two hemispherical endcaps are presented. These objects are made of stainless steel and filled with air. Their radius is 27 mm and their radius ratio b/a is 0.97 (b is the inner radius and a the outer radius). The direction of insonification is parallel to the main object axis. The reduced frequency range $k_1 a$ of the study varies from 10 to 40. These experimental results are explained by writing the stationary condition on the large circumference of objects, in the meridian plane. This condition depends on the phase velocity in the length of the cylindrical shell and in the spherical shell. These two velocities are computed separately in relation to the frequency and used to apply the stationary condition. The obtained results are in good agreement with the experimental study.

3:30

9PA10. Radiation of a guided wave at the end of the cylindrical shell. Fernand Léon, Dominique Décultot, Florence Lecroq, and Gérard

Acoustic scattering from an isotropic elastic hollow cylindrical shell of infinite length excited by an obliquely incident plane acoustic wave is investigated. The waves generated at an incident angle α on a determined area of a cylindrical shell immersed in water and filled with air, propagate in a direction parallel to the cylinder axis and re-emit their energy during the propagation at an angle α . The various velocities of propagation of helical waves allows one to record separately the resonance spectrum of each type of waves. Then, the radiation, in the far field, can be observed for each wave, at the end of a shell of semi-infinite length, with a quasiharmonic method. In this case, no phenomenon of stationary waves can occur in the cylinder length. The axial displacement is analyzed and the use of the Rayleigh integral allows one to determine a model of the pressure radiated by the circular section of the tube. The results of the theoretical calculation are in agreement with the results of experiments.

9PA12. Measurement of periodic surface backscattering to high diffractive order. H. R. Suiter (Code 2120, Coastal Systems Sta., Panama City, FL 32407-5000)

New tank measurements have been done of the backscattering from two periods each of a pressure release sine wave, Stokes wave, and wedge profile grating from about 60 to 200 kHz. The rms surface deviation of all six targets was selected to be approximately the same value at 2.3 mm. The measurements were done beyond the fifth diffraction order, and in one case, to the eighth order using a new technique. Dispersion curves were followed almost exactly except for some anomalous dispersion attributed to Wood *P*-type anomalies. Comparison of reflection coefficients is made to a modified Kirchhoff scattering model that takes into account the bounded nature of the beam. Agreement is good. [Work supported by ONR.]

3:45

9PA11. Acoustic scattering from multiple rigid cylinders. Angie Sarkissian (Naval Res. Lab., Washington, DC 20375-5000)

Results are presented for acoustic scattering from multiple rigid, axisymmetric cylinders where all the cylinders have a common axis of symmetry with an acoustic field incident from an arbitrary direction. The boundary-element method is implemented to compute the scattered field from cylinders with hemispherical endcaps where the azimuthal dependence is separated first to reduce computation time. Scattering computations are made for various two- and three-cylinder configurations where the dependence of the scattered field on the distance between the cylinders is examined as well as scattering from a large cylinder in the presence of a smaller cylinder.

4:15

9PA13. The effect of irregularity on the scattering of acoustic waves from a ribbed plate. Douglas Photiadis (Naval Res. Lab., Code 5131, Washington, DC 20375)

The scattered field from an irregularly ribbed fluid-loaded plate is "measured" using numerical simulations. The diffuse field scattered by the irregularity exhibits strong coherent features in the far-field scattering pattern associated with phase matching to locally propagating Bloch waves on the structure. These features are also apparent in the scattering from finite periodic arrays, but are strongly enhanced by irregularity. Some examples of the behavior of the system are presented and the scaling of the cross section with the number of scatterers and irregularity is determined.

FRIDAY AFTERNOON, 15 MAY 1992

SALON E, 1:00 TO 4:50 P.M.

Session 9SP

Speech Communication: Consonant Perception and Production

Gail R. Tomiak, Chair

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Chair's Introduction—1:00

Contributed Papers

1:05

9SP1. Effects of place of articulation and vocalic context on the perception of VOT continua in French and English. Terrance M. Nearey and Bernard L. Rochet (Dept. of Linguistics, Univ. of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Previous findings indicate that VOT varies in production with place of articulation and with following vowel, though details differ somewhat in French and English. Here, perceptual results are presented involving VOT continua ranging from -80 to $+80$ ms VOT in 10-ms steps.

Twelve different continua were constructed (three places of articulation by four vowels). Seven English and six French speakers categorized these continua as /b,d,g,p,t,k/ and simultaneously provided "goodness" ratings. Voiced-voiceless crossover boundaries for English speakers showed highly significant main effects and interactions for both factors. Differences among the French speakers were not significant. The "goodness" rating data of the French speakers, however, did show significant effects for place, but none for vowel context. Although vowel effects in this production data are about twice as large in French as in English, French listeners show less evidence of sensitivity to vowel context. Pos-

sible reasons for this lack of parallelism will be discussed. [Work supported by SSHRC.]

1:20

9SP2. Use of vocalic cues to final consonant voicing in perception and production of English CVC's by native speakers of Arabic. Court Crowther and Virginia Mann (Dept. of Cognitive Sci., Univ. of California, Irvine, Irvine, CA 92717)

Native Arabic speakers who were either late learners (first resided in an English speaking country after high school) or early learners (first resided during high school) were compared in two experiments that tested their use of vocalic duration and *F1* offset frequency as final consonant voicing cues in the English CVC's "pod" and "pot." Experiment 1 measured both vocalic duration and *F1* offset frequency of subjects' productions of these syllables; and experiment 2 considered their perception of tokens from synthetic "pod"-"pot" continua that varied vocalic duration and *F1* offset. The results indicate some effects of native language experience and age of learning. Use of vocalic duration in production was greater for early than late learners, although considerably smaller than that of native English speakers. The early learners showed native-like sensitivity to vocalic duration as a voicing cue, but the late learners did not. The range of *F1* offset frequency differences tended to be smaller than in native productions, and to show less effect of voicing. However, both nonnative groups showed native-like sensitivity to *F1* offset.

1:35

9SP3. Stop consonant identification using auditory images. Lawrence Feth, Robert Allen Fox, and Ina Bicknell (Div. of Speech and Hear. Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002)

A number of researchers over the past decade have argued that salient characteristics of auditory processing must be incorporated into speech recognition models before they can accurately represent human speech processing which might, in turn, improve recognition accuracy. Recently, Fox and Feth [Proc. Ninth International Symposium on Hearing, Carcans, France (1991)] processed a series of stop + vowel CVs (using [bdgptk] and the vowels [i e æ o u]) through Patterson and Holdsworth's auditory sensation processing (ASP) model producing a set of "stabilized auditory images." Each image represented a spectral representation of 15 ms of the acoustic signal and the images for each token could be displayed in rapid sequential order producing a "cartoon" of the auditory changes evident over a wide range of auditory analysis channels. A set of viewing experiments conducted using three subjects demonstrated that these stops (and vowels) could be accurately identified on the basis of the dynamic auditory cues contained in the auditory images. Accuracy rates up to 99% for voicing and 92% for place of articulation were obtained. The present study is a continuation of this research using a longer display window that may more accurately reflect short auditory memory in perceptual processing. [Research supported, in part, by an AFOSR grant to L. Feth and a NIA grant to R. Fox.]

1:50

9SP4. Acoustic properties contributing to the classification of place of articulation for stops. T. V. Ananthapadmanabha (Voice and Speech Systems, Temple St., Malleswaram, Bangalore, India, 560 003) and Kenneth N. Stevens (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

The production of stop consonants generates several kinds of acoustic properties: (i) the spectrum of the initial transient and burst indicating the size of the cavity anterior to the constriction; (ii) place-dependent articulatory dynamics leading to different time courses of the

noise burst, onset of glottal vibrations and formant transitions; (iii) formant transitions indicating the changing vocal tract shape from the closed position of the stop to a more open configuration of the following vowel. The interest in this study is to measure the relative contributions of these acoustic properties to the classification of the consonantal place of articulation using a semi-automatic procedure. The acoustic data consisted of a number of repetitions of voiceless unaspirated stops in meaningful words spoken by several female and male speakers. The spectra averaged over the stop release and at the vowel onset were used as the acoustic feature. Speaker-independent and vowel-independent classification was about 80% using either the burst or vowel onset spectrum and a combined strategy led to a higher accuracy. Studies with additional acoustic properties that relate to articulatory dynamics, such as VOT and formant transitions, are planned.

2:05

9SP5. An analysis of invariance in English stop consonants. Kevin H. Richardson (Modern Technologies Corp., San Antonio, TX 78214)

Speech researchers have been attempting to isolate acoustic invariance in the speech signal for the past several decades. Currently, however, no conception of invariance has proven adequate to the task of demonstrating the existence of invariant acoustic features in the six English stop consonants. In the present study, static and dynamic versions of two recent invariance proposals [Sawusch and Dutton (in press) and Forrest, Weismer, Milenkovic, and Dougall (1988)] were developed and tested on a corpus of 1392 naturally produced stop and stop consonant cluster tokens. Classification accuracies of the metrics using stepwise discriminant analyses showed performance to be significantly different from chance but lower than expected had the metrics isolated invariance in the signal. Results will be discussed in terms of general human auditory-phonetic coding ability. Also, the notion that invariant acoustic features are the most useful way of conceptualizing the problem of phoneme recognition will be questioned. [Work supported by NIDCD Grant No. R01-DC00219 to SUNY at Buffalo.]

2:20

9SP6. A signal detection theory analysis of phoneme boundary shifts. Richard Fahey (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

Phoneme boundary shifts are well documented in the speech perception literature. Signal detection theory (SDT) models of speech perception allow for two possible explanations for these. Boundary shifts may be due to criterion shifts at a decision level or to changes in the underlying representations of stimuli. SDT methods can be applied to both identification and fixed-level discrimination data. While not providing a definitive explanation of a phoneme boundary shift, the results of such analyses can constrain explanations. In the present investigation, SDT analyses were applied to both identification and fixed-level discrimination results in which phoneme boundary shifts were observed. These data come from two sets of studies: (1) a "rate normalization" experiment, in which the duration of syllables affected voicing judgments, and (2) a trading relations study, in which the medial fricative/affricate boundary was influenced by both the silent duration before the fricative noise and fricative rise time. Implications for existing explanations of both of these phoneme boundary shifts are discussed. [Work supported by NIH.]

2:35

9SP7. Mismatch negativity evoked potential to categorically perceived speech stimuli. Anu Sharma, Nina Kraus, Therese J. McGee, Thomas D. Carrell, and Trent G. Nicol (Dept. of Commun. Sci. and Disord., Audiology, 2299 Sheridan Rd., Frances Searle Bldg., Evanston, IL 60208)

Categorical perception of certain speech and speech-like sounds has been critical to models of speech perception for decades. Furthermore, refining our knowledge of the stimulus and task characteristics that lead to categorical perception continues to be an important part of research in speech perception. Auditory evoked potentials (AEPs) have the capability of providing new perspectives on this important phenomenon. The mismatch negativity (MMN) AEP is elicited in response to a physically deviant stimulus occurring in a series of standard stimuli [Näätänen *et al.*, *Acta Psychol.* **42**, 313–329 (1978)]. The MMN has been shown to be sensitive to fine acoustic differences and was used in the present study to examine a stimulus continuum that typically leads to categorical perception. The primary auditory cortex is thought to be a major source contributing to the generation of the MMN. In the

present experiment a stimulus continuum ranging from /da/ to /ga/ was synthesized by varying the frequencies of F_2 and F_3 . An identification task showed the expected steep slope in the identification function along this continuum, indicating that they were perceived categorically. An electrophysiologic index of discrimination was obtained by recording the MMN in adults to stimulus pairs, having equal acoustic differences, within and across categories. The MMN was observed in all subjects and was the same size both across and within categories. That is, the MMN indicated equal discrimination both across and within categories. These results suggest that the MMN reflects the processing of auditory aspects of these speech-like signals, but not processing into phonetic categories.

2:50–3:05

Break

3:05

9SP8. Acoustic-perceptual factors in nasal place assimilation. Patrice Speeter Beddor and David K. Evans-Romaine (Program in Linguistics, Univ. of Michigan, Ann Arbor, MI 48109)

Phonologically, nasal consonants often assimilate to the place of a following oral stop. While the source of this assimilation is generally taken to be anticipatory coarticulation, this study examines the possibility that acoustic-perceptual factors are partially or even primarily responsible. Two American English speakers produced /zVNCə/ tokens (V = /ɪ/, /æ/, /ə/; NC = /mb/, /nd/, /ŋg/). Stimuli were split into /zVN/ and /Cə/ portions. /zVN/ portions whose final N was identified with at least 97% accuracy (four judges) were cross-spliced with /Cə/ portions to yield all possible N–C pairings. Listeners identified N and C of the cross-spliced /zVNCə/ stimuli. Homorganic clusters were correctly identified 95% of the time (preliminary results with eight listeners). Listeners varied in accuracy of N identification in non-homorganic clusters, but across subjects one-third of nonhomorganic nasals were misidentified, with homorganic responses comprising 80% of these errors. Although different places of articulation showed different response patterns, overall results indicate that place cues for nasals may be overridden by those of a following oral stop. Even nasals whose place is unambiguous in final position may shift perceptually before heterosyllabic oral stops.

3:20

9SP9. The role of allophonic variation in the perception of the junctural position of /l/. Dawn L. Dutton (Dept. of Psychol., SUNY at Buffalo, Amherst, NY 14260)

Many theories of spoken word recognition assume speech is segmented at syllable or word boundaries prior to contact with the lexicon. A number of researchers [see K. W. Church, *Cognition* **25**, 53–69 (1987)] have observed systematic acoustic differences (or allophonic variations) within a phonetic class that may serve as cues to juncture points between syllables or words. The most common difference cited between syllable initial and syllable final /l/ is the proximity of the first two formants. Some researchers have also noted a difference in the amplitude profile of /l/ depending on its position relative to a word boundary. In this experiment, two starting phrases, “see leaves” and “seal eaves,” were synthesized. F_1 frequency profile, F_2 frequency profile, and amplitude profile were varied orthogonally for both starting phrases. The junctural position of /l/ was cued primarily by F_1 fre-

quency profile. F_2 frequency profile, and thus the proximity of the first two formants, and amplitude profile seemed to play a very small role in the perception of /l/ position. [Work supported by NIDCD Grant No. DC 00219 to SUNY at Buffalo.]

3:35

9SP10. Visual-auditory integration and vocalic effects on fricative perception. Naoyuki Takagi and Virginia Mann (Dept. of Cognitive Sci., Univ. of California at Irvine, Irvine, CA 92717)

This study investigates the effect of visual presentation of four syllables ([sa], [ʃa], [su], [ʃu]) on perception of a fricative noise continuum. The visual stimuli were taken from videotapes of a speaker uttering the four syllables. The nine-step noise continuum ranged from [s] to [ʃ], and when followed by the acoustic vocalic portions taken from natural utterances of [sa], etc., replicated Mann and Repp’s (1980) findings about the context effects of vowel lip rounding and consonant place of articulation on fricative perception. To determine if similar context effects would occur when the vocalic information was specified visually rather than auditorily, stimuli from the noise continuum were dubbed in place of the natural utterances on the video tapes and subjects identified the visual-auditory hybrids by naming each syllable aloud. The results showed a significant lip rounding effect on perception of “s” vs “sh” indicating that listeners’ compensation for the decrease in frication noise frequency that is caused by anticipatory lip rounding takes place whether the lip rounding is perceived visually or auditorily. The effect of consonant place of articulation was less consistent.

3:50

9SP11. Integration of acoustic cues for consonants across frequency bands. B. A. Rubin, R. M. Uchanski, and L. D. Braida (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

To what extent can disjoint frequency bands of speech be treated as independent perceptual channels when consonants are identified? To test the independence hypothesis, consonants were filtered into four bands (0.0–0.7, 0.7–1.4, 1.4–2.8, and 2.8–5.0 kHz). Filtered speech bands were presented to listeners with normal hearing monotonically. Pairs of these bands (e.g., 0.0–0.7 and 0.7–1.4 kHz) were presented dichotically, and summed (producing a 0.0–1.4 kHz band) for diotic presentation. Consonant identification scores for the combined band

conditions were predicted by models of cue integration that have been applied to audiovisual consonant identification [L. D. Braid, Q. J. Exp. Psychol. 43A, 647-677] on the basis of confusions observed in the single band conditions. For most subjects and conditions, diotic and dichotic scores were similar. For a few subjects scores were lower for the diotic 0.0-1.4 kHz band case than for the corresponding dichotic condition. Predicted two-band scores agreed more closely with dichotic than diotic scores. Thus while most listeners appear to integrate cues across bands efficiently, some apparently experience cross-band interference of peripheral origin.

4:05

9SP12. Functional organization of velar movements following jaw perturbation. H. Betty Kollia, Vincent L. Gracco, and Katherine S. Harris (Haskins Labs., 270 Crown St., New Haven, CT 06510)

Mechanical perturbation of the articulators has been used to examine motor control principles underlying speech production. One common observation is that if one member of a group of functionally related articulators is perturbed during articulation, the other members compensate. The compensatory response patterns reflect sensorimotor organization for speech, and thus have implications for speech production theories. The present experiment examines such linkages by analyzing the spatiotemporal response patterns to jaw perturbation displayed by the anatomically linked and remote jaw, upper lip, lower lip, and velum. Subjects were fitted with a mandibular prosthesis to allow delivery of jaw-lowering, random (20% of the trials) perturbations during productions of the utterance /mabnab/. Lip, jaw, and velar kinematics were recorded optoelectronically and simultaneously with the acoustic signal. These movements exhibited compensatory actions including increased movement displacement and oral closing velocity. Movement duration changes were also observed. Coordinative timing among lips, jaw, and velum was generally maintained following the perturbation. Variations in that timing appear to depend on the temporal relations between the onset of the perturbation and the onset of the component speech event. [Work supported by NIH Grant Nos. DC-00121, and DC-00594 to Haskins Laboratories.]

4:20

9SP13. The role of remote rate information and relative acoustic resemblance on phonetic representations. R. S. Newman and J. R.

Sawusch (Dept. of Psychol., SUNY at Buffalo, Amherst, NY 14260)

Miller and Liberman [Percept. Psychophys. 25, 457-465 (1979)] demonstrated that listeners use vowel duration information when making the /ba/-/wa/ distinction. They also found that adding a final consonant to the syllables shifted the /ba/-/wa/ boundary. However, they did not examine the effects of changing the final consonant duration. Furthermore, by using stimuli that were voiced throughout, the potential effects of acoustic and auditory similarity between the initial consonant distinction and the succeeding phonemes could not be examined. In this experiment CVC syllables were used to examine the effect of this later-occurring speaking rate information on an initial /sh/-/ch/ distinction. The syllables ended with /s/, and both vowel and final fricative duration were varied independently. This allowed one to examine the effect of acoustic similarity on the coding of rate information. The relevance of these results to the auditory and/or phonetic coding of speaking rate information will be discussed. [Work supported by NIDCD Grant No. DC00219 to SUNY at Buffalo.]

4:35

9SP14. The effects of coarticulation on Japanese voiceless consonant discrimination. Hideki Kojima, Shinta Kimura, and K. H. Loken-Kim (Speech Processing Sect., Advanced Systems Res. Div., Fujitsu Labs., 1015, Kamikodanaka, Nakahara-Ku, Kawasaki, 211 Japan)

In this paper, the results of a study of automatic discrimination of Japanese voiceless fricatives in vowel (/a,i,u,e,o,n/)-consonant (/F,s,h/) -vowel (/a,i,u,e,o/) syllables are presented, and the effects of coarticulation on this discrimination are discussed. A total of 4032 speech data uttered by 16 male subjects were used for the study. A higher recognition rate was obtained when the acoustic features were extracted from the consonant and succeeding vowel transition region than when they were extracted from the consonant region or the succeeding vowel transition region alone. Significantly, the extracted features from the consonant-vowel transition region were most effective in the recognition of consonants succeeded by /a/, but less effective for those succeeded by /i/, /e/, and /o/. Furthermore, when /i/ and /e/ were preceded by /F/ and /s/ the recognition rate was low. Similarly when /o/ was preceded by /F/ and /h/ the recognition rate was low. This phenomenon can be explained from the perspective of consonant-vowel coarticulation.