

## PROGRAM OF

### The 110th Meeting of the Acoustical Society of America

Hyatt Regency Hotel • Nashville, Tennessee • 4–8 November 1985

TUESDAY MORNING, 5 NOVEMBER 1985

SUITE 5, 8:00 A.M. TO 12 NOON

#### Session A. Underwater Acoustics I: Air–Sea Interaction and Noise

Robert W. Farwell, Chairman

*Naval Ocean Research & Development Activity, Code 113, NSTL Station, Mississippi 39529*

Chairman's Introduction—8:00

#### *Invited Papers*

8:05

**A1. The effects of sea surface roughness on the transmission of sound through an air–ocean interface.** J. E. Barger and D. A. Sachs (Bolt Beranek and Newman Inc., 10 Moulton Street, Cambridge, MA 02238)

This is a review of recent experimental and theoretical studies of the influences of the sea surface on the transmission of airborne sound into the ocean. Most applications will depend upon conditions where the refracted grazing angle is greater than approximately  $30^\circ$  since the loss is very large for smaller angles. For grazing angles larger than  $30^\circ$ , two parameters serve fairly well to characterize the transmission loss through the sea surface. The first is a roughness parameter  $R$  which is equal to the sea wave height measured in airborne sound-speed wavelengths. The second is a diffraction parameter  $DL$  which is equal to the Fresnel zone diameter measured in correlation lengths of the sea surface slopes. When  $R$  is less than unity and  $DL$  is less than 3, there is usually no effect due to surface roughness. When either condition is not satisfied, there is usually more transmission loss than there would be through a flat surface. Exceptions to this depend upon the refracted grazing angle and upon the angle between the wave crests and the incident sound.

8:30

**A2. Review of source spectrum level density of wind-generated noise.** James H. Wilson (Wilson's Arctic Research, Inc., 16333 Keeler Drive, Granada Hills, CA 91344)

The source spectrum level density of wind-generated noise has been deduced from ocean measurements, laboratory experiments, and theoretical studies in the past 25 years. Selected contributions are reviewed and the results are compared. Since the source level results compare favorably, at least in deep water, an average source spectrum level density as a function of frequency and surface wind speed is proposed for use in noise models. The differences between the dipole and "rocking" dipole source directivity patterns is discussed and wind-generated noise modeling issues are addressed. There seems to be little agreement on the underlying physical mechanisms of wind-generated noise. The question of what information would be added to the source spectrum level densities if a theory for the underlying physical mechanisms were known is also addressed.

8:55

**A3. Low-frequency ocean surface noise sources.** W. M. Carey (Naval Ocean R&D Activity, NSTL, MS 39529) and M. P. Bradley (Planning Systems Inc., Slidell, LA 70458)

The interaction of the wind with the ocean surface has long been recognized as the major source of ocean ambient noise. High-frequency noise data (200 to 2 kHz) has consistently been found to have strong wind-dependent characteristics associated with spray, splashes, bubbles, and rain. Recently, wave–wave interaction has been shown to be a source of infrasonic (0.2 to 2 Hz) noise and ocean bottom microseisms. Generally, the low-frequency noise (2 to 20 Hz) is associated with noise from distant ships. However, narrow-band (as opposed to 1/3-octave) measurements show in addition to the noise from ships a wind-dependent characteristic. Furthermore, mid-ocean basin vertical directionality measurements show noise intensity near the horizontal with a broad frequency characteristic in diverse geographic locations. These results suggest a wind-generated noise due to a mechanism such as wave–wave interaction, wind turbulence, or the interaction of surface waves with turbulence is coupled into the mid-basin sound channel by either a shallowing sound channel such as found at high latitudes or a down-slope conversion process due to the basin boundaries and sea mounts. Theoretical

expressions are derived from first principles following the approaches of Yen and Perrone [Naval Underwater Systems Center TR5833 (1979)] and Huon Li [Naval Ocean R&D Activity, TN89 (1981)] yielding the frequency-dependent radiation characteristics for wave-wave interaction, wind turbulence, and wave-turbulence interaction. These results show that wave-turbulence interaction is a possible source of wind-driven noise in the 10- to 200-Hz regions. Other possible mechanisms such as nonlinear capillary wave interactions are discussed and compared to this wave-turbulence mechanism.

9:20

**A4. Vertical directionality of ambient noise—A review.** W. von Winkle and D. Browning (Naval Underwater Systems Center, New London, CT 06320)

The vertical directionality of ambient noise in the sea falls into three frequency regimes: 0–200 Hz, dominated by long range sources; 200–10 000 Hz, relatively small directionality; 10 kHz and above, downward rays and heavily attenuated. A review is made of data mechanisms and our present prediction capability.

## Contributed Papers

9:45

**A5. The air/sea interface as a wave source.** J. E. Ffowcs-Williams (Engineering Department, Cambridge University, Trumpington Street, Cambridge CB2 1PZ, England)

The surface waves on the ocean are coupled to the underwater sound field, the coupling arising either because the one generated the other, or because they were both created coherently by a common source. This paper discusses the various mechanisms of sound generation at and near the ocean surface and identifies parameters controlling the regimes where the various sources are dominant. Reference will be made to the results of some exact model problems involving aerial turbulence-induced surface waves and their associated sound field in an infinitely deep ocean. These model problems utilize the techniques of aero-acoustics on the ocean sound problem and indicate that some aspects have not previously been adequately modeled when the surface is vigorously disturbed in the main noise-producing conditions.

10:00

**A6. On the wind-wave interaction mechanism.** R. H. Mellen (PSI Marine Sciences, New London, CT 16320) and D. Middleton (127 E. 91 Street, New York, NY 10128)

Sonar and radar backscattering measurements indicate that the wind-driven sea surface does not behave according to linear theory at high wavenumbers. Backscatter strengths at small grazing angles are much greater than expected and the scatterers appear to move without the usual dispersion associated with "capillary" waves. Theory shows that nonlinear effects of surface-drift and wind-wave interactions can account for the shocklike properties. Nonlinear wind-wave effects have been observed in wave-flume measurements by spectral analysis of wave-gauge data. Initially periodic ripples develop coherent harmonics with increasing fetch; however, subharmonic growth leads to rapid degeneration and a continuous spectrum. The mechanism appears to involve intermodulation between wind and wave leading to chaotic behavior and redirecting the energy cascade to lower wavenumbers. In the equilibrium stage, further development of longer gravity waves ceases as they outrun the source, incoming energy being balanced by high-wavenumber dissipation. Unstable disturbances generated on the surface should degenerate into ensembles of solitons, explaining the Doppler spreads and backscatter cross sections observed at high frequencies.

10:15

**A7. Infrasonic ambient ocean noise: Northeast Pacific Ocean.** Rudolph H. Nichols (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Measurements of ocean ambient noise were made at three widely separated deep-water bottom locations in the N. E. Pacific, at eight frequen-

cies in the range from 2.5–20.0 Hz, for 40 consecutive days. Concurrent data on wind speed and wave height were collected. Analysis indicates that the spectrum level of infrasonic noise is linearly related to the log of the wind speed above a threshold level. There is evidence that the noise can be directly associated with the wind, rather than through the surface waves it produces. [Work supported by ONR.]

10:30

**A8. Underwater noise caused by precipitation.** Joseph A. Scrimger (Jasco Research, Ltd., 9865 West Saanich Road, Sidney, British Columbia, Canada V8L 3S1)

The characteristics of underwater noise in the ocean generated by precipitation are important to weather forecasters and oceanographers since they permit the detection and measurement of rain over the ocean by remote (i.e., buoyed or bottom-mounted) acoustic sensors. We have recently observed the character of the underwater noise generated by rain, hail, and snow. The spectrum of rain noise, for wind speeds below 1.5 m/s, shows a peak of 13.5 kHz with a sharp cutoff on the low-frequency side and a gradual falloff (7 dB per octave) on the high-frequency side. Stronger winds smear the peak. Hail spectra show a peak at 3.0 kHz with a gradual (roughly 11 dB per octave) falloff on both sides. The spectrum of snow noise is unique. Our instrumentation permitted the measurement of the drop (or stone) size distributions in the precipitation. These findings will enhance the art of remote acoustic sensing in oceanography. [Work supported by: Supply & Services Canada; Institute of Ocean Sciences, Sidney, BC, Canada; Atmospheric Environment Service, Toronto, Canada; and Department of National Defence, Canada.]

10:45

**A9. Bubble-related ambient noise in the ocean.** A. Prosperetti (Department of Mechanical Engineering, The Johns Hopkins University, Baltimore, MD 21218)

Several mechanisms by which bubbles can contribute to ambient noise in the ocean are described and their effectiveness estimated. At low frequency, up to a few tens of Hz, bubbles are driven into oscillation by oceanic turbulence. The normal quadrupole radiation mechanism of turbulence thus acquires a monopole character. At frequencies from around 1 to a few kHz single bubbles formed by breaking waves radiate in free oscillation. In the range of hundreds of Hz the acoustic emission may be due to collective oscillations of systems composed of many bubbles. The efficiency of all these mechanisms is estimated on the basis of an adaptation of Lighthill's theory of aerodynamic noise. Finally, at frequencies above several kHz, drop impact and free oscillations of bubbles thereby produced appear to be responsible for the ambient noise. [Work supported by SACLANT ASWR, La Spezia, Italy.]

11:00

**A10. Ambient noise directionality at two locations in the Gulf of Mexico.** Ronald A. Wagstaff (Naval Ocean Research and Development Activity, Code 245, NSTL, MS 39529)

Low-frequency ambient noise directionality measurements were made by a towed line array in the Gulf of Mexico at two different sites. The first site was seaward of the continental shelf and north of the Yucatan Peninsula. The second site was near the entrance to the Gulf of Mexico. The presence of a shallow shelf at the first site effectively shields the array from long-range acoustic propagation from the south. This shielding, combined with favorable propagation conditions and high-density shipping to the north, is responsible for the high degree of anisotropy in the measured ambient noise horizontal directionality pattern at the first site. The measured ambient noise directionality pattern at the second site shows the strong influence of the shipping entering and exiting the Gulf of Mexico. Because the acoustic propagation conditions and the shipping in the Gulf of Mexico are approximately repeatable annually, these two directionality measurements are considered to be good estimates for the ambient noise for future years. A discussion of the horizontal directionality as well as additional ambient noise statistics for the two sites will be given.

11:15

**A11. Computer simulation of the vertical structure of mid-ocean ambient noise.** Donald A. Murphy, Donald R. Del Balzo, and Ronald A. Wagstaff (Naval Ocean Research and Development Activity, Ocean Acoustics Division, Code 240, NSTL, MS 39529)

The acoustic field for a realistic distribution of noise sources was calculated at a deep-water location in the Northeast Pacific corresponding to the CONTRAK VI measurement site [M. Z. Laurence and D. J. Ramsdale, *J. Acoust. Soc. Am. Suppl.* 1 77, S70 (1985)]. The calculations were made with the NORDA Parabolic Equation programs along four horizontal directions representing distinctly different environments and historical shipping distributions. The calculated field at the CONTRAK VI site was sampled by vertical line arrays to determine the vertical arrival angles of the noise. This vertical structure was studied as a function of array depth, array length, and frequency for each horizontal direction and compared with the CONTRAK VI measurements. The shape and

depths of the resulting vertical noise notch are heavily affected by arrivals from sources over continental slopes and in cold water. [Work supported by DARPA.]

11:30

**A12. Calculations of ocean ambient noise over a continental slope.** D. R. Del Balzo, M. J. Authement, and C. T. Mire (Naval Ocean Research and Development Activity, Ocean Acoustics Division, Code 240, NSTL, MS 39529)

Computer calculations of slope effects on surface noise at 100 Hz were made using the NORDA high-angle, implicit finite difference acoustic field model. A geoacoustic description of a continental margin, which is typical of the east coast of the U. S., was chosen for analysis (vertical slope of 2.8 deg and covered by 500 m of silt). In order to enhance the effect of bottom interaction, a Summer Sargasso Sea sound-speed profile was used. The calculations indicate the presence of a deep noise notch over the slope (for certain basin and shelf noise generating locations) which is deepest near the ocean surface (25 dB) and which decays with depth. The potential for signal-to-noise ratio gain is suggested. [Work supported by NAV-AIR.]

11:45

**A13. Arctic Ocean basin theory.** Ira Dyer (Department of Ocean Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Low-frequency ambient noise sources observed in the Arctic are discretely located in space and time. With a large number of events per unit area per unit time, the total number of contributions to any one observation grows with horizontal range but also shrinks in intensity inversely with range. This standoff therefore suggests that limits on observed ambient noise are set by basin bathymetry and/or by sound absorption. In one useful model we find the source spectral density typical of low-frequency events is changed via the basin by an  $f^2$  shaping at frequencies below, and by an  $f^{-1}$  shaping at frequencies above a characteristic frequency. [Work supported by ONR.]

TUESDAY MORNING, 5 NOVEMBER 1985

DAVIDSON ROOM A, 8:15 TO 11:50 A.M.

## Session B. Noise I: Noise in the Workplace—James H. Botsford Memorial Session

John J. Earshen, Chairman  
*Metrosonics, Inc., P. O. Box 23075, Rochester, New York 14692*

### Invited Papers

8:15

**B1. A tribute to James H. Botsford (1925–1984) and a historical review of impulse noise measurement in the steel industry.** Edwin H. Toothman (Occupational Health and Safety Division, Bethlehem Steel Corporation, Bethlehem, PA 18016-7699)

A tribute to Jim Botsford who passed away on 18 May 1984 will be presented. A historical review of noise measurement in the steel industry, including impulse noise, highlighting Jim's endeavors will be given. The instrumentation and methods used for noise measurements along with rating information will be discussed. Problems associated with impulse noise measurement will be reviewed and questions regarding proper instrumentation, analysis, and rating for impulse noise will be raised.

**B2. Classification of impulse noise: Comparison of crest factor and kurtosis as metrics.** John Erdreich (National Institute for Occupational Safety and Health, Cincinnati, OH 45226)

In this session honoring Jim Botsford, I will address one aspect of a problem with which he was concerned, whether impulsive noise should be (or could be) considered separately from continuous noise exposures. Recently, I presented a method for classifying noise as impulsive based on a statistical distribution measure of "peakedness," the kurtosis ( $b_2$ ). This is the first step in addressing the "separability" of noise types. It is likely that Botsford and others of a practical demeanor would ask what advantage, if any, this has over familiar measures such as crest factor (cf). By defining a ratio,  $\alpha$ , such that  $\alpha = b_2/\text{cf}^2$ , we show that the measures are related such that for a given impulse form,  $\alpha$  is constant for any duration signal. Only the form of the impulse changes the value of  $\alpha$ . Thus, for a rectangular impulse,  $\alpha = 1$  and for an exponentially decaying sinusoid,  $\alpha = 3/2$ . The crest factor may, therefore, have some utility for classifying impulsive environments. This utility is diminished, however, for cases at the border between impulsive and continuous noise. For signals with  $3 < b_2 < 6$ , the crest factor is unreliable for classifying the environment because of its dependence on a single peak. Examples of the relative efficiency of the metrics for classification will be drawn from actual samples of industrial noise.

9:05

**B3. Noise dosimeter transient response characteristics.** John J. Earshen (Metrosonics, Inc., P. O. Box 23075, Rochester, NY 14692)

Noise dosimeters used to monitor worker exposures to noise must have prescribed transient response characteristics to satisfy the requirements of the U. S. Department of Labor, Occupational Safety and Health Administration. In addition, computation of noise dose or average sound level must be based on a 5 dB per time doubled trading ratio. Various individuals have concluded that there is an apparent discrepancy between measured and computed doses when industrial noise contains impulsive components. The discrepancy results when computation neglects the effects of the prescribed transient response characteristics. In particular, "slow" response has a critical effect when trading ratios of 4 or 5 dB are used. A dosimeter is required to produce measurement results which correspond to results that can be computed from SLM readings. This paper shows the relationship between transient response and measured dose resulting from noise containing impulses. Furthermore, experimental results are presented which show that dosimeter and SLM derived results are consistent and correct. Reported results purporting to show that dosimeters give incorrect measurements of impulsive noise are shown to be incorrectly evaluated because theoretically computed reference values fail to account for U.S. D.O.L. OSHA prescribed transient response. The effect of substituting "fast" response is also presented.

9:30

**B4. Assessing the hazards of impulse noise superimposed on background noise.** D. Henderson and R. P. Hamernik (Callier Center of the University of Texas at Dallas, 1966 Inwood, Dallas, TX 75235)

Impulsive noise, either in the military or industrial setting, rarely occurs without some substantial background noise. This paper reviews a series of animal experiments which show synergistic interactions between relatively "safe" continuous and impulsive noise. In all the experiments, chinchillas were either exposed to (1) 60 min of background noise (either 2-4 kHz or 0.5-1 kHz) at levels ranging from 95 to 89 dB; (2) 50 spark discharge-generated impulses with peak sound pressure levels ranging from 158 to 137 dB and  $A$  durations of 32 to 64  $\mu$ s; (3) various combinations of (1) and (2). The animals' hearing was measured immediately after the exposure, and at regular intervals over a 30-day period. After 30 days, the cochleas were analyzed for losses of sensory cells. The results show that certain combinations of individually "safe" impulse and continuous noise can (1) produce a synergistic interaction; (2) that these interactions behave systematically as either the level of the impulse or the continuous noise is reduced; and (3) that the interaction effect is enhanced as the frequency spectra of the two noises begin to overlap. The results of the experiments will be discussed in terms of the mechanisms of noise-induced hearing, implications for measurement of noise environments loss, and the implications for public health standards. [Work supported by NIOSH.]

9:55

**B5. Hearing protector attenuation as influenced by frequency and time variation of the noise.** Daniel L. Johnson and Myron D. Smith (Larson-Davis Laboratories, 280 S. Main, Pleasant Grove, UT 84062)

Over a decade ago, Jim Botsford proposed two innovative concepts that provide the basis of this study. First, he proposed correcting the attenuation of hearing protectors by a "C-A" adjustment [J. H. Botsford, Sound Vib., 32-33 (Nov. 1973)]. Using three separate time history dosimeters, set to 5-s periods, the C-weighted and A-weighted levels at the shoulder were measured; also, the level under the muff was recorded. The results show that most of the time the C-weighting outside the muff is the best predictor of the A-weighted level under the muff when used with a single value of noise reduction. Another concept of Botsford was the TTS meter, in which the intermittency pattern of a noise could be assessed [J. H. Botsford, Am. Ind. Hyg. Assoc. J. 32, 92-95 (1971)]. Harris and the first author further enhanced this concept by providing a TTS-based intermittency correction factor to adjust a noise dose [D. L. Johnson and C. S. Harris, Proc. Noise Expo, Chicago, IL, 16-122 (1979)]. Using this adjusted dose, our current data indicate better performance by muff-type hearing protectors than simply calculating inside and outside average noise levels might predict. We also noted that a person's



## Contributed Papers

10:20

**B6. Development of a new artificial head for impulse noise attenuation measurements of hearing protective devices.** Christian Giguere, Hans Kunov, Sharon M. Abel, Juergen Schroeter, Glenn Willmott, and Barry S. Myers (Institute of Biomedical Engineering, University of Toronto, Toronto, Canada M5S 1A4, and Silverman Hearing Research Laboratory, Mount Sinai Hospital, 600 University Avenue, Toronto, Canada M5G 1X5)

While psychoacoustical methods to evaluate the sound attenuation of hearing protective devices have been standardized (CSA Z94.2-M1984; ASA STD 1-1975) and proved to be practical for continuous noise, no standard exists for impulse noise. Microphone-based techniques seem to provide the best avenue for the measurement of impulse noise attenuation at the present time. This paper presents our progress in the design of a new artificial head simulating the relevant mechanical and acoustical parameters of the human head influencing the protected and unprotected eardrum sound pressure. These parameters include: (1) the diffraction of sound by the human head, (2) the mechanical impedance of human tissues, (3) the circumaural contours, (4) the external ear features such as pinna, ear canal, and eardrum impedance, and (5) the vibration of the head in the sound field. [Work supported by the Defence and Civil Institute of Environmental Medicine, Canada.]

10:35

**B7. The dependence of critical level upon duration.** W. Dixon Ward and Dorisann Henderson (Hearing Research Laboratory, University of Minnesota, 2630 University Ave. SE, Minneapolis, MN 55414)

In the chinchilla, the critical level for single exposures to 700- to 2800-Hz noise of 20 min or longer duration has been found to be between 112 and 120 dB SPL. For example, a 22-min exposure at 120 dB produced considerably more permanent threshold shift, PTS (50 dB at 2000 Hz), and histological damage [80% destruction of the outer hair cells (OHC)] than a 220-min exposure at 112 dB or any single uninterrupted exposure longer than 220 min but equivalent in energy to the 220-min 112-dB exposure (20 dB PTS, 10% OHC destruction). However, in the present experiment, daily exposure for 9 weeks, Monday through Friday, to 40 7.2-s bursts of 120-dB noise spread over an 8-h "workday" (hence with 12 min between bursts) produced negligible PTS and OHC destruction. Clearly, therefore, the critical level depends on duration and temporal pattern, and so regulatory systems governing exposure of workers to industrial noise that put a fixed ceiling on exposure levels regardless of temporal pattern—whether this ceiling be 115 dBA, 130 dBA, or 140 dB peak—will often, if not usually, be incorrect. [Research supported by NIH Grant 12125.]

10:50

**B8. The effect of impulse intensity and the number of impulses on hearing and cochlear pathology in the chinchilla.** James H. Patterson, Jr., Ilia M. Lomba-Gautier, Dennis L. Curd (Sensory Research Division, U. S. Army Aeromedical Research Laboratory, Fort Rucker, AL 36362-5000), Roger P. Hamernik, Richard J. Salvi, C. E. Hargett, Jr., and George Turrentine (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

Forty-one chinchillas, divided into seven groups, were exposed to 1, 10, or 100 noise impulses having peak intensities of 131, 135, 139, or 147 dB. Hearing thresholds were measured in each animal prior to exposure using an avoidance conditioning procedure. Threshold shifts were moni-

tored at regular intervals over a 30-day post-exposure period. A surface preparation of the cochlear sensory epithelia was performed approximately 90 days after exposure. There was generally an orderly relation between the amount of permanent threshold shift and the severity of exposure, and a general agreement between averaged histological data and the audiometric data. Detailed relations between temporary and permanent threshold shift, cochlear pathology, and exposure variables are discussed, as are the implications of these data to the development of exposure criteria.

11:05

**B9. Impulse noise repetition rate: Implications for the equal energy hypothesis.** D. Henderson, R. P. Hamernik, and R. J. Salvi (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

The equal energy hypothesis (EEH) for impulse noise is based on the assumption that the hearing hazard associated with an impulse noise exposure is a monotonic function of the total A-weighted energy received by a listener. A series of experiments will be described in which the EEH was evaluated using exposure paradigms in which the repetition rate (4/s to 1 every 16 s) and the intensity (107- to 131-dB peak SPL) of the impact noise were varied. Each of the various exposure paradigms had the same total energy. If the magnitude of the hearing loss is determined by the amount of acoustic energy, then all groups should develop the same amount of hearing loss. For equal intensities, the degree of damage increases with increased repetition rate. For example, the level of TTS increased approximately 25 dB when the repetition rate was increased from 1 every 16 s to 4/s. The effect of increasing the repetition rate also resulted in greater PTS and cochlear damage. The results are discussed in terms of the EEH and its appropriateness to impulse/impact noise. [This work was supported by NIOSH-R010H1152.]

11:20

**B10. Temporary functional changes in men due to exposure combinations of noise and low-frequency vertical vibrations.** Olavi J. Manninen (The Academy of Finland, c/o Department of Public Health, University of Tampere, P. O. Box 607, 33101 Tampere, Finland)

The main purpose of this study was to produce additional information on the connections between the temporary hearing thresholds, body upright posture sway, and cardiovascular changes induced by complex exposure conditions. Seven healthy male students were exposed consecutively five times to mere noise, to mere whole body vibration, and simultaneously to noise and vibration at 20 °C. The noise categories were: (1) no noise and (2) noise of 90 dBA. The categories of low-frequency whole body vibration (Z axis) were: (1) no vibration, (2) vibration within the range 4.4–5.6 Hz, (3) vibration within the range 2.8–5.6 Hz, (4) vibration within the range 2.8–11.2 Hz, (5) vibration within the range 1.4–11.2 Hz, and (6) sinusoidal vibration with a frequency of 5 Hz. The (rms) acceleration in all the vibration models was 2.12 m/s<sup>2</sup>. The TTS<sub>2</sub> values at 4 and 6 kHz increased as a result of simultaneous exposure to noise and vibration significantly more than as a result of exposure to noise alone. The means of the sway variances in the X and Y directions at 0.1 Hz and within the range 0.06–2.00 Hz increased only when the vibration in the noise-vibration combination was a sinusoidal one. The changes in the heart rate, R-wave amplitude, blood pressure, and the haemodynamic index scores also depended on the bandwidth of the vibration, the number of consecutive exposures, and on whether the subjects were simultaneously exposed to noise in addition to vibration. [Work supported by The Academy of Finland.]

**B11. A characterization of room noise levels at telephone locations and the effect upon employees' work and communication.** George R. Auten, Jr. (United States Air Force, HQ SAC/DERI, Offutt AFB, NE 68113-5000), Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910), and Richard G. Pearson (Department of Industrial Engineering, North Carolina State University, Raleigh, NC 27695-7906)

Noise surveys were conducted at telephone locations in four types of work environments: State government office buildings, fast-food restaurants, automobile dealers/maintenance shops, and shopping mall retail

stores. Two types of surveys were taken. A type I survey was conducted during normal working conditions and a type II survey was conducted at selected sites in order to better define the dominant noise sources. Acoustic measurements made included A-weighted, C-weighted, F-weighted (flat), and octave-band sound pressure levels. General room descriptors were also noted for each survey location. A total of 270 telephone locations were surveyed at 51 sites. The employees were asked to assess their work place with respect to noise and its effect on their work, face-to-face, and telephone communication activities. Summary statistics for each type of sound survey and the responses from the questionnaires given out during the surveys will be presented.

TUESDAY MORNING, 5 NOVEMBER 1985

REGENCY BALLROOM IV, 8:30 TO 11:45 A.M.

### Session C. Speech Communication I: Synthetic Speech and Its Evaluation

Mark Y. Liberman, Chairman

AT&T Bell Laboratories, Murray Hill, New Jersey 07974

#### Contributed Papers

8:30

**C1. Text to speech—An overview.** J. P. Olive and M. Y. Liberman (AT&T Bell Laboratories, Murray Hill, NJ 07974)

We discuss the design of a text-to-speech synthesizer, which accepts any type of English text as input, and creates an appropriate speech signal as output. Effective algorithms for converting text to sound must make use of intermediate data structures that systematically encode the degrees of freedom available to speakers of the language being synthesized. These data structures are an engineering approximation to what linguists call *phonological representations*; we will call them "*P-structures*." Any TTS system must: (1) define its version of *P-structures*; (2) design and implement algorithms for transforming input text into *P-structures*; (3) design and implement algorithms for transforming *P-structures* into sound. Task #1 is mainly a problem in applied linguistics; task #2 can best be seen as applied AI; task #3 is an application of phonetics and signal processing expertise. Because of the interdependence of approximate solutions in different portions of the system, integration of the various parts is a non-trivial problem. We will analyze an example of a TTS system, showing how these problems were handled in building and combining its numerous pieces. Finally, we point out some areas where better solutions are needed, and suggest how to find such solutions.

8:45

**C2. A hybrid domain articulatory speech synthesizer.** Juergen Schroeter and Man Mohan Sondhi (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Articulatory speech synthesizers model the human vocal tract by means of its geometrical and functional properties. It is believed that this approach can be advantageous in speech coding at bit rates below 4800 b/s. Existing articulatory synthesizers work in the time domain. They either solve a system of differential equations for the vocal tract and the glottis, or synthesize speech using wave digital filters. The first approach is computationally very cumbersome. Both approaches have difficulties in incorporating important acoustic parameters, for example, the radiation impedance at the lips, wall vibration, and other losses. So far a realistic glottis model suitable for the wave digital filter approach does not exist. We have combined a nonlinear time-domain model of the vocal cords [K. Ishizaka and J. L. Flanagan, *Bell Syst. Tech. J.* **51**, 1233-1268 (1972)] with a linear frequency-domain chain-matrix model of the vocal tract [e.g., M. M. Sondhi, paper #4.5.1, *Proc. Int. Congress on Acoustics*, Paris,

France, 1983, Vol. 4, pp. 167-170]. The interface between these two models consists of convolving the glottal flow in the time domain with impulse responses of the tract obtained by inverse FFT. Examples of synthesized speech using manually generated and measured tract areas will be given.

9:00

**C3. Towards developing a parametric model of dynamics of LPC parameters.** Bishnu S. Atal (Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

Acoustic parameters, such as LPC parameters, vary continuously as functions of time. The underlying speech (phonetic) events, however, are discrete. How do we go from continuous acoustic representation to discrete phonetic representation? We will discuss in this paper a parametric model for representing variations of LPC parameters with time. We define a matrix  $Y$  to express the LPC parameter values at successive time instants with different columns representing time. We then write matrix  $Y$  as a product of three matrices  $A$ ,  $Z$ , and  $P$ : a central speech information matrix  $P$  representing underlying speech events, a left matrix  $A$  relating the speech events to the LPC parameters, and a right matrix  $Z$  representing the spreading of speech events along the time axis. The number of columns in the matrix  $Z$  is equal to the number of speech events in a given speech segment and is much smaller than the number of time frames in that segment. We assume that there is overlap between the spreading functions of only neighboring speech events, that is, the matrix  $PP'$  is a tridiagonal matrix. We will discuss procedures to determine the matrices  $A$ ,  $Z$ , and  $P$  such that the mean-squared error between the elements of the matrix  $Y$  and the product  $AZP$  is minimized. We will also discuss conditions under which the selection of the matrices is unique.

9:15

**C4. Aligning phonemes with the corresponding orthography in a word.** S. G. C. Lawrence and G. Kaye (IBM UK Scientific Centre, St. Clement Street, Winchester, SO23 9DR, England)

As part of a speech synthesis project a technique has been developed for aligning the phonemes in a phonemic transcription with the graphemes in a word. For example *creationism* /kri:ənɪzəm/ can be aligned as *c.r.e.a.t.i.o.n.i.s.m* with /k.r.i:ɪ.ɛɪ.j.n.i.z.ə.m/. The technique has been used to determine in a machine-readable dictionary, where an entry is

given with multiple orthographies and one pronunciation, whether the pronunciation may be used for all the orthographies. For example, in the Collins English Dictionary there appears the entry *abutment* /əˈbʌtmənt/ or *abuttal*. We know these two words are synonymous but are not pronounced the same. *a.b.u.tt* aligns with /ə.b.ʌ.t/ but the *al* cannot be aligned with /mənt/. The pronunciation of *abutment* cannot be used as the pronunciation of *abuttal*. The aligned forms of words and their pronunciations, together with statistics on frequency of rule use, is seen as an invaluable aid for developing a set of text-to-phoneme rules, where the ordering of rules, based on frequency of occurrence, is critical. It may also be useful as part of a pattern matching process in a speech recognition system. This paper will describe the algorithm and will give some results of using it against a phonemically tagged version of the Lancaster-Oslo/Bergen Corpus.

9:30

**C5. A dictionary-intensive letter-to-sound program.** Cecil H. Coker (Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

We treat the letter-to-sound problem as that of deriving all regularly inflected forms of English words from a dictionary of base forms. A computer program must perform all lexical operations a human might have to perform in using an abridged desk dictionary. Of course this includes addition of "s," "es," "d," "ed," "er," and "ing," with their accompanying sound changes, deletions of mute "e," and changes from "y" to "i." The program must also handle other forms. Stress-neutral endings like "less" and "ness" are quite simple. Four other types of endings, exemplified by "ation," "ity," "ian," and "ency," are more complex—reordering stress within the base word, and causing vowel shifts and consonant sound change. The problems with these endings are much like those for straight letter-to-sound rules, but with pronunciations of base form and ending as available information. With prefixes and compounds, the relative stress of constituent parts will depend on the intended part of speech for the target word, and on the properties of the parts. Final vowels of Greek prefixes may or may not reduce to schwa, depending on properties of the base forms they modify. With a programmed dictionary driver, we are currently finding ~99.9% of words in the Brown corpus (excluding proper names and misspellings)—~75% from exact dictionary entries, and the rest by derivation. We default to Mark Liberman's Name-Say program for words not found or derived.

9:45

**C6. Stress assignment in letter-to-sound rules for speech synthesis.** Kenneth W. Church (AT&T Bell Laboratories, Murray Hill, NJ 07974)

This paper will describe a program that attempts to determine word stress from spelling. Four sources of constraints will be discussed: (1) syllable weight, (2) part of speech, (3) morphology, and (4) etymology. Stress assignment is a well-established weak point for many speech synthesizers because stress dependencies cannot be determined locally. It is impossible to determine the stress of a word by looking through a five- or six-character window, as many speech synthesizers do. Well-known examples such as *dégrâde/dégradation* and *télégraph/télégraphy* demonstrate that stress dependencies can span over two and three syllables. In addition, examples such as *calculi* and *tortoni* demonstrate that stress is sensitive to etymology, a nonlocal property of the entire word. *Calculi*, like most other latinate forms, obeys the main stress rule of English, which assigns antepenultimate stress to nouns ending in two light syllables. In contrast, *tortoni*, like most other Italian and Japanese loan words, receives penultimate stress. Stressing Italian and Japanese surnames with the main stress rule of English as some commercial synthesizers do, leads to unacceptable results, as will be demonstrated. Our program produces much more acceptable results by estimating the etymology (by counting three letter tri-grams) and then adjusting the stress rules appropriately.

10:00

**C7. Speech synthesis of Japanese from written Esperanto.** Shigeru Sato (Tohoku Institute of Technology, Yagiya-kasumicho 35-1, Sendai, 982 Japan) and Hideki Kasuya (Department of Electronics, Utsunomiya University, Utsunomiya, 321 Japan)

With a subsystem designed to extract meaning from written Esperanto being added to the speech synthesis system of Japanese generated in the semantic/syntactic base [S. Sato and H. Kasuya, J. Acoust. Soc. Am. Suppl. 1 73, S5 (1983)], we present an overview of this translation-synthesis system including a knowledge base editor for implementing linguistic information needed. Allowing a wide range of data modification and file manipulation operations, the frame editor can edit on a frame/slot basis the rules and dictionaries used in parsing Esperanto and generating Japanese. The main portion of the knowledge base is occupied with descriptions about case structures, selectional restrictions, hierarchical relations among items, etc. for detecting anomaly and ambiguity of a given Esperanto sentence. The semantics gives to the meaning a case frame expression, from which the sentence generator produces a surface form with some trimming with phonological and accentuation rules. The synthesis program further converts the surface into acoustic parameters, rendering a time series of control parameters for the terminal analog synthesizer.

10:15

**C8. Text-to-speech synthesis of Japanese based on the Cepstrum-CV method.** Yasushi Ishikawa and Kunio Nakajima (Mitsubishi Electric Corp., Information Systems and Electronics Development Laboratory, 325 Kamimachiya, Kamakura, 247 Japan)

A text-to-speech synthesis system for Japanese is described. In synthesis of Japanese, generation of fundamental frequency pattern is one of the most important problems. We proposed the prosody rules that generate intonation pattern of any Japanese text using syntactic information. The prosody rules consist of modification rules, syntax rules, and a generation rule. Connecting several words, the system makes an accent phrase and gives it an accent using the dictionary and the rules (modification rules). The parameters used in a subsequent generation rule are modified on the basis of the syntactic structure of the input text (syntax rules). Then, a fundamental frequency pattern is generated from these parameters (generation rule). In this system, the synthesizer parameter is Cepstrum. Parameter rules assign segment duration and concatenate CV syllables Cepstra. Thus, this system translates any text of Japanese written with the syntactic symbols into good quality and natural speech.

10:30

**C9. Fast sequential closed-phase glottal inverse filtering based on optimally weighted recursive least squares.** T. C. Luk (Baldwin Technology Corporation, 5118 S. Dansher Road, Countryside, IL 60525) and J. R. Deller, Jr. (Northeastern University, Department of Electrical & Computer Engineering, and The Center for Speech Processing & Perception, Boston, MA 02115)

Presented in this paper is the theoretical basis, with simulation verification, for a sequential method of deconvolution of the glottal waveform from voiced speech. The technique is based upon a linear predictive model of the vocal tract, and the assumption of a "pseudo-closed phase" (PCP) (noisy closed phase) of the glottis during each pitch period. Existing techniques for closed phase glottal inverse filtering (CPIF) employ "batch"-type methods which are generally slow, highly user interactive, and restricted to the use of one cycle of data in the analysis. The basic ideas underlying CPIF, and a brief review of existing methods, will be presented in the first part of the paper. In the second part of this paper, requisite theoretical results and the new method will be developed. In particular, these will include a unified theory of CPIF in which the selection of closed phase points is viewed as a data weighting process. This viewpoint readily admits the use of more than one cycle of data in the analysis (advantageous

when the data are noisy), and further leads to the use of optimal weighting of the accepted data. The theory of "membership set" identification is used as a basis for optimization of weights. Novel weighting strategies employed in a conventional recursive least-squares algorithm form the basis of the improved technique. The last part of the paper contains simulation studies and computational considerations. The new method is shown to result in significant increases in both accuracy and computational efficiency.

10:45

**C10. A backward-type band-split adaptive predictive coding system with dynamic bit allocation for wideband speech and sound signals.** Shinji Hayashi, Masaaki Honda, and Nobuhiko Kitawaki (NTT Musashino Electrical Communication Laboratories, 3-9-11 Midori-Cho Musashino-Shi, Tokyo 180, Japan)

This paper proposes a new backward-type speech coding system (ADPCM-AB) for wideband (7 kHz) speech and sound signals. In this system, a split-band adaptive predictive coding scheme with gradient PARCOR lattice filter and a dynamic bit allocation scheme are employed, where quantization bits are dynamically allocated over the subbands (frequency), the subintervals (time), and the channels (stereo) in accordance with the distribution of the residual energies. They serve to remove the redundancies due to the periodic concentration of the prediction residual energy and the nonuniform nature of the speech spectrum. The ADPCM-AB needs neither longer delay time than 4 ms nor transmission of side information parameters because the parameters for predictive coding and dynamic bit allocation are calculated with the locally decoded signals. It is clarified that the ADPCM-AB system has the best speech quality among the conventional backward-type coding systems. It is also shown that this system provides speech quality subjectively equivalent to 11-bit linear PCM (176 kb/s) at 64 kb/s.

11:00

**C11. Objective quality measures applied to enhanced speech.** John H. Hansen and Mark A. Clements (School of Electrical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Previously reported experiments have indicated that computer enhancement of single channel speech in wideband noise generally does not result in increased intelligibility. Our research goal has been to investigate the effect of such enhancement on speech quality. Our experimental framework employed natural speech degraded by additive white, non-white, and aircraft cockpit noise over a wide range of signal-to-noise ratios (— 15 to 20 dB). An iterative enhancement technique based on a method reported by Lim and Oppenheim, was generalized to allow nonwhite additive noise. Spectral estimates of the noise were made using maximum-likelihood method, maximum-entropy method, Burg's method, periodogram, Bartlett's method, and Pisarenko harmonic decomposition. Quality was estimated using a variety of objective measures which have been shown to correlate well with subjective quality scores resulting from the diagnostic acceptability measure (DAM). Over a large number of noise conditions the best objective qualities resulted when Bartlett, Burg, and

maximum-entropy spectra were used. In general, significant improvements in terms of objective measures were observed when enhancement took place. [Work supported by Lockheed Georgia Company.]

11:15

**C12. Use of DP matching for evaluating synthesized speech.** Takashi Saito, Yasuhiro Matsuda, and Toyohisa Kaneko (Science Institute, IBM Japan, Ltd., 5-19 sanbancho, chiyodaku, Tokyo, Japan 102)

We examined the applicability of DP matching as a tool for evaluating methods for connecting synthesis units. In speech synthesis by rule, speech quality depends greatly on how to connect synthesis units (e.g., syllables). The most widely used method is to use human ears. Although it is justifiable as the final judgment, it is time-consuming as well as fairly listener-dependent. The method proposed here is to use the amount of residual errors of DP matching, which has been widely known in speech recognition. When we confront the problem of selecting parameters of a particular connecting method, we record words naturally spoken (presumably continuous) and then operate DP matching between the natural speech and synthetic speech. The parameter set that produces the least amount of residual error is selected. We devised DP matching particularly suited to this purpose. We found that the optimized parameters with this method agreed well with those obtained with human ears and believe that this proposed automated method is applicable as a substitute of human ears for sizable classes of evaluation problems.

11:30

**C13. Evaluation of a two-microphone speech-enhancement system.** M. V. McConnell, P. M. Zurek, P. M. Peterson, and W. M. Rabinowitz (Research Laboratory of Electronics, Room 36-736, Massachusetts Institute of Technology, Cambridge, MA 02139)

Two variants of a processing scheme first proposed by Kaiser and David [J. Acoust. Soc. Am. 32, 918 (1960)] were evaluated for enhanced intelligibility. In both systems the running cross correlation between two microphone signals is used to derive a time-varying gating signal that multiplies the sum of the microphone signals to form a monaural stimulus. With a target source straight ahead and an interference source off axis, this processing results in an output that is increasingly attenuated with decreasing target-to-interference ratio. In the first system this processing was applied to the wideband inputs and in the second it was applied independently in four octave bands. With the latter system three degrees of processing (determined by the function relating output attenuation to cross-correlation) were investigated. Intelligibility tests were conducted using sentences as the straight-ahead target and continuous discourse as the off-axis (90° azimuth) interference. Contrary to expectations based on the report of Kaiser and David, in no case was speech intelligibility improved as a result of processing. Apparently, the benefits of removing remote interference (temporal and spectral) do not outweigh degradations of the target signal. [Work supported by NIH.]

**Session D. Architectural Acoustics I: Recording Studio Control Room Acoustics**

Russell E. Berger, Co-Chairman

*Joiner-Pelton-Rose, Inc., 4125 Centurion Way, Dallas, Texas 75234*

Ewart A. Wetherill, Co-Chairman

*Bolt Beranek and Newman Inc., 21120 Vanowen Street, Canoga Park, California 91304****Invited Papers*****9:00****D1. Optimizing control rooms for stereo imagery.** Douglas R. Jones, William L. Martens, and Gary S. Kendall (Northwestern Computer Music, School of Music, Northwestern University, Evanston, IL 60201)

Control rooms designers typically measure and specify rooms according to their physical structure and acoustic properties. They are unable, however, to measure or predict how well the room will support the subjective qualities of stereo imagery produced over loudspeakers. As the quality and salience of stereo imagery improve through the use of more sophisticated recording and processing techniques, control room requirements become more stringent. Beyond speaker placement, there are three primary factors that influence the perception of stereo images: time-energy-frequency characteristics of the speakers, spatio-temporal distribution of early reflections, and the inclusion of acoustic diffraction. These are easily measured through the use of time delay spectrometry (TDS), but at present an adequate model for predicting subjective response from these physical measurements is lacking. Ensuring the perception of optimal stereo imagery requires the application of standardized subjective evaluation techniques. Currently under development at Northwestern Computer Music (NCM) is an evaluation technique using the Listening Environment Diagnostic Recording (LEDR<sup>TM</sup>), which enables the immediate assessment of changes in stereo imagery that result from progressive changes in control room acoustical treatment. Field tests indicate that LEDR<sup>TM</sup> is valuable in the design and modification of control rooms for optimizing stereo imagery.

**9:30****D2. Recording control room design incorporating a reflection-free zone and reflection phase grating acoustical diffusors.** Peter D'Antonio, John H. Konert (RPG Diffusor Systems, Inc., 12003 Wimbledon Street, Largo, MD 20772), and Charles Bilello (Master Sound Astoria, 258 Fairlawn Avenue, West Hempstead, NY 11552)

The acoustical diffusing properties of flat, mono, and bicylindrical surfaces, alternating absorptive and reflective panels, and a wide variety of reflection phase grating diffusors (RPG) will be discussed. Experimental TEF measurements including energy-time, energy-frequency, time-energy-frequency, and polar angle-energy-frequency curves plus theoretical Kirchhoff modeling calculations will be presented. A new control room design incorporating a reflection-free zone (RFZ), in the front half of the room, and a diffuse zone, in the rear half of the room created with RPG diffusors, will be described and documented. The RFZ minimizes the speaker boundary interference over a wide volume surrounding the mix position. The RPGs provide a diffuse sound field which enhances the perception of spatial textures and helps maintain the stereo perspective across the entire width of the mixing console and in the rear of the room. The RFZ/RPG design minimizes frequency coloration, image shifting, and provides accurate stereo imaging.

**10:00****D3. Optimization of monitoring signal reflection patterns in recording studio control rooms.** J. M. Wrightson (Joiner-Pelton-Rose, Inc., 4125 Centurion Way, Dallas, TX 75244) and Brad S. Brubaker (Department of Psychology, University of Wisconsin, Milwaukee, WI 53201)

Recent control room design practice has often included rear/side wall reflective surfaces to provide a discrete, delayed reflection of the direct monitor speaker signal. This delayed signal is typically directed to the ear contralateral from the direct sound. To assess the perceptual effects of this practice, listeners were presented with pulses, speech, and music to both ears with a delayed repetition to a single ear. The level and delay duration of the monaural signal were varied. Under some conditions, 50 additional channels of delay at low intensity were presented diotically to simulate room reflections. Listeners were asked to make judgments of detectability for the monaural signal and to describe any lateralization changes in the undelayed signal when differences were heard. These conditions were repeated in a control room environment. The results of these judgments indicate that addition of the monaural delayed signal is easily detected for most conditions and substantially alters the spatial perception of the undelayed signal.

10:30

**D4. Reverberation time in physically small rooms.** Don Davis (Syn-Aud-Con, P. O. Box 669, San Juan Capistrano, CA 92693) and E. T. Patronis, Jr. (School of Physics, Georgia Institute of Technology, Atlanta, GA 30332)

Physically small rooms are often erroneously analyzed and measured according to the classical statistical acoustical equations and techniques. Evidence is presented herein to support the position that a truly diffuse reverberant field does not exist in such spaces at a level above that of the ambient noise.

11:00

**D5. Isolating music and mechanical equipment sound sources with gypsum board partition systems.** H. Stanley Roller (United States Gypsum Company, Architectural & Construction Services Department, 101 South Wacker Drive, Chicago, IL 60606)

The 100-Hz low-frequency limit for sound transmission loss measurements and the STC rating system have seriously limited the development of a vocabulary of practical lightweight constructions which can effectively isolate music and mechanical equipment sound sources. This paper discusses the results of recent research on gypsum board partition systems which include sound transmission loss measurements down to 50 Hz and the evaluation of a number of design factors such as balanced and unbalanced constructions, panel damping, cavity resonances, and cavity insulation. Measurements made at 50-, 63-, and 80-Hz one-third octave bands are in good agreement with calculated TL at these frequencies. These data, plus calculated TL at 31 and 40 Hz, are used to further evaluate the proposed MTC (music and mechanical equipment transmission class) rating system which is based on 125- to 5000-Hz data.

11:30

**D6. Review of some approaches to architectural-acoustic design of sound recording facilities in the U. S. S. R.** Gregory A. Kacherovich (Jaffe Acoustics, Inc., 114A Washington Street, Norwalk, CT 06854)

Rooms are classified for their purpose in relation to acoustics, based on the authors's experience in architectural-acoustic design and consulting in the U. S. S. R. for over 20 years. General acoustic requirements are set for each group of rooms. Auditoria are classified depending on the sound source origin. Special attention is brought to sound recording studios and related spaces used for radio, television, and motion picture facilities. General acoustic criteria, general approach in design, selection of size, shape, and materials are discussed for different groups of rooms including motion picture and television stages, music and speech recording studios, re-recording studios, control rooms, etc. Music recording studios are considered to be of two major types depending on the use of distant and close pickup microphones. Several multi-purpose studios were designed and built in the 1970's providing the necessary acoustic environment for music and speech recording as well as for re-recording purposes. Systems for variable acoustics were developed providing a wide range of variability.

TUESDAY MORNING, 5 NOVEMBER 1985

SUITE 4, 9:00 A.M. TO 12 NOON

### **Session E. Physical Acoustics I: Astrophysical Applications of Acoustics**

**Ilene J. Busch-Vishniac, Co-Chairman**

*Department of Mechanical Engineering, University of Texas, Austin, Texas 78712*

**Ethan T. Vishniac, Co-Chairman**

*Department of Astronomy, University of Texas, Austin, Texas 78712*

#### ***Invited Papers***

9:00

**E1. Sound in the early universe.** Ethan T. Vishniac (Department of Astronomy, University of Texas, Austin, TX 78712)

The standard cosmological model is based on the approximation that the universe is homogeneous and isotropic. The overwhelming success that this model has enjoyed is based on its ability to predict the primordial abundance of elements and the isotropy of the microwave background. However, the universe itself is not homogeneous. Any realistic model must include the perturbations from homogeneity that evolved into galaxies and galaxy clusters. Perturbations on these scales behaved like sound waves at sufficiently early times when the gas in the universe was almost completely ionized. In this talk I will review the work done on the linear and

nonlinear evolution of these sound waves. The isotropy of the microwave background places firm limits on the amplitude of the sound waves at the epoch of recombination. As a result we are able to rule out the hypothesis that the universe consists entirely of radiation and ordinary matter.

9:30

**E2. Shock waves in astrophysics.** Edmund Bertschinger (Department of Astronomy, University of California, Berkeley, CA 94720)

Gravitational forces acting on a large scale often generate supersonic velocities in cosmic gas leading to shock waves. Astrophysical shock waves range in size from the earth's bow shock in the solar wind to interstellar shock waves generated by supernova explosions, and perhaps to galaxy-size explosions in the early universe. Shock waves are important to astronomers and astrophysicists for a variety of reasons. Shocked gas emits radiation which serves as a useful diagnostic of the composition and state of the gas. Radiative shock waves, because they can greatly increase the gas density and hence gravitational binding energy, are also important in enhancing star formation. A review will be given of the causes, occurrence, and significance of astrophysical shock waves. The theoretical, observational, and computational methods used to study them will also be discussed.

10:00

**E3. Physical mechanisms in stellar pulsations.** Arthur N. Cox (Los Alamos National Laboratory, University of California, Los Alamos, NM 87545)

Stars evolve from their births to their deaths by converting their store of hydrogen to helium, and then much of this helium is fused to heavier elements such as carbon, oxygen, and up to iron. During this evolution, the stellar mass may decrease by a stellar wind mass loss, the radius usually greatly increases, and the radiation luminosity emitted at the surface grows until the stellar death results in a very small compact object. During this evolution, there are often several stages when the structure of a star is unstable against pulsations. These pulsations are observable and indicators of the internal details of its structure. Six of these self-excitation mechanisms, which produce limited amplitude pulsations, will be discussed and demonstrated. Three deep-seated ones are the modulated nuclear fusion reactions at the stellar centers, the possible Kelvin-Helmholtz instability at the surface of a rapidly rotating core of the star, and oscillation of convective eddies which has a restoring force due to a composition gradient in deep layers. Three mechanisms which operate in the outer layers are the oscillations of convective eddies restrained by a strong magnetic field, and the thermodynamic effects of blocking and hiding of the radiation luminosity due to the ionization of the abundant elements, hydrogen and helium.

10:30

**E4. Coherent vortical features in a turbulent two-dimensional flow and the Great Red Spot of Jupiter.** Philip S. Marcus (Division of Applied Sciences and Department of Astronomy, Harvard University, Cambridge, MA 02138)

We present the results of an initial-value study of a nearly inviscid flow in a low aspect ratio, rapidly rotating, cylindrical annulus with a free upper surface and with a sloping bottom surface. In the limit of very rapid rotation, the equations and solutions of this flow are the same as those of a planetary atmosphere whose density decreases exponentially with height. We have found numerically that for a wide range of initial conditions the flow settles into a statistically steady state that consists of an isolated coherent spot of vorticity superposed on a turbulent zonal (statistically axisymmetric) flow whose time-averaged vorticity is linear in radius (or linear in latitude of a planetary atmosphere). The strength and sign of the vorticity in the zonal flow and in the spot, the shape and location of the spot, and the interaction of the spot with other features are quantitatively similar to the properties of the Red Spot of Jupiter and the zonal wind in which it is located. The time-averaged features of the zone and the spot are surprisingly insensitive to a number of different types of boundary conditions. We present examples of solutions where two or more spots (present as initial conditions) merge into one spot, and where a spot with the wrong sign of vorticity breaks up and reforms rotating with the correct sign. An analytic model is developed based on the numerical results that shows that under some conditions a spot must necessarily form, and that two or more spots must merge into a single spot. The model's predictions of the size, shape, strength, and location of the spot are in good agreement with the numerical results. We conclude by discussing the feasibility of reproducing this flow (i.e., a coherent spot in a turbulent background) in the laboratory.

11:00

**E5. Gravitational waves: A new window for astronomy.** Peter F. Michelson (Physics Department, Stanford University, Stanford, CA 94305)

Gravitational waves were predicted more than 50 years ago by Einstein as a consequence of the general theory of relativity. Because of the weakness of the gravitational interaction, efforts to directly detect gravitational waves have focused on astrophysical sources rather than terrestrial sources. In laboratories around the

world, second-generation cryogenic acoustic detectors and laser interferometric detectors are being developed with sensitivity and bandwidth sufficient not only to verify directly the existence of gravitational waves, but also to study the received waveforms. Because the gravitational radiation emitted by an astrophysical source contains information about the source that is orthogonal to information obtained from electromagnetic signals, the direct detection of gravitational waves will open a new window for astronomy. In this review, the likely astrophysical sources of gravitational radiation and the technology of second-generation acoustic detectors will be discussed.

11:30

**E6. Vibrations in the environment of large future telescopes.** E. T. Pearson (National Optical Astronomy Observatories, Advanced Development Program, 950 North Cherry Avenue, Tucson, AZ 85726)

As larger telescopes are designed with more servo-controlled elements, the chance of undesirable vibrations increases. What are the sources of the vibrations and how can the telescope be designed to avoid problems? If problems can be created by acoustical and mechanical vibrations, might there also be a use for input vibrations?

**TUESDAY MORNING, 5 NOVEMBER 1985**

**REGENCY BALLROOM II AND III, 9:00 TO 11:30 A.M.**

## **Session F. Physiological and Psychological Acoustics I: Acoustics and Mechanics of the Auditory Periphery**

**George F. Kuhn, Chairman**

*Vibrasound Research Corporation, 10957 East Bethany Drive, Aurora, Colorado 80014*

### ***Contributed Papers***

**9:00**

**F1. Some physical insights and mathematical models for the sound reception by the human pinna.** George F. Kuhn (Vibrasound Research Corporation, 10957 East Bethany Drive, Aurora, CO 80014)

The frequency response, directional gain, standing wave ratios, and pressure distributions for models of the human pinna, composed of regular geometrical sections, have been determined analytically to approximately 12 kHz. These analytical models yield new, significant insights into the physics of sound collection by the pinna, its efficiency, and the effect of the pinna's geometrical details on the frequency response and the directivity. Some simplifying approximations of the mathematical formulations yield useful rules about the relationships between the pinna geometry and the gain function as well as the spatial and spectral locations of pressure-peaks and pressure-nulls. [Work supported by NIH/NINCDS.]

**9:15**

**F2. Acoustics of ear canal measurement of eardrum SPL.** Samuel Gilman and Donald Dirks (Head and Neck Surgery Department, Center for the Health Sciences, UCLA School of Medicine, Los Angeles, CA 90024)

This paper reports on a study to determine the effects of standing acoustic waves on the accuracy of ear canal determination of eardrum SPL. The study was conducted in three parts: (1) A theoretical analysis of standing waves of sound pressure in a lossless transmission line having the geometry of the average ear canal with terminations corresponding to the extremes in normal eardrum impedance, (2) actual probe measurements on ANSI S3.25-1979 ear simulators, modified so that ear canal length and eardrum impedance could be varied, (3) probe determination of eardrum SPL on real ears. Results from simulator and real ear measurements agree with the theoretical determinations obtained from the standing wave study. Theoretical and actual standing wave ratios agree with published real ear data [E.A.G. Shaw, "Transformation of sound pressure level from the free field to the eardrum in the horizontal plane," J. Acoust. Soc. Am. 56, 1848-1861 (1974)].

**9:30**

**F3. Measurement and specification of the human ear canal geometry.** Michael R. Stinson (Division of Physics, National Research Council, Ottawa, Canada K1A 0R6)

Above 8 kHz the sound pressure transformation along the human ear canal (a central issue in the development of high-frequency audiometry) depends very much on the details of the canal geometry. In addition to canal volume and length, which provide an adequate description at lower frequencies, one must account for the curvature of the ear canal and the variation of cross-sectional area along the canal length. In particular, calculation of the sound pressure transformation requires specification of cross-sectional areas perpendicular to a curved axis that follows the "center" of the canal. A photogrammetric technique has been developed that accurately determines the geometry of human ear canals and specifies it in this format. Stereoscopic photographs of cadaver ear canal castings are analyzed to define a large number (up to 1000) of coordinate points on the surface of each casting. A numerical procedure extracts the curved, center axis that passes through the points and calculates cross sections normal to this axis. The geometry of one ear canal has been successfully obtained using the technique, and measurements on several more are under way.

**9:45**

**F4. Acoustic pressure distributions in cadaver ear canals.** George F. Kuhn (Vibrasound Research Corporation, 10957 East Bethany Drive, Aurora, CO 80014)

The plane wave acoustic pressure distributions in cadaver ear canals have been calculated up to 20 kHz, taking into account the geometry of the ear canals. Also, the pressure distributions across the eardrums as well as the average—and central—pressure (at the umbo) have been calculated. The results indicate that at high frequencies the spatially averaged pressure across the eardrum, the central pressure (at the umbo), and the maximum pressure in the standing wave in a uniform extension of the ear canal differ little from each other. Since the pressure distribution across the eardrum varies greatly in amplitude and phase, a rotational movement



about the umbo or central portion of the eardrum is shown to be a potential mechanism of transduction of sound to the middle ear. [This research is sponsored by NIH/NINCDS.]

10:00

**F5. A model for changes in middle-ear transmission caused by stapedius-muscle contractions.** Xiao-Dong Pang and William T. Peake (Room 36-825, Research Laboratory of Electronics and Department of Electrical Engineering & Computer Science, MIT, Cambridge, MA 02139 and Eaton-Peabody Laboratory, Massachusetts Eye & Ear Infirmary, Boston, MA 02114)

We have observed in cats that when the stapedius muscle contracts, the stapes moves by as much as  $60\mu\text{m}$ , while the incus and malleus are not detectably displaced ( $<1\mu\text{m}$ ). This observation provides physiological support for the hypothesis that stapedius contractions change only the impedance of the stapes. A network model has been developed to test this hypothesis. The middle ear is represented as a linear two-port network that is loaded by the impedance of the stapes and cochlea. A quantitative description of the two-port is derived from measurements of (a) acoustic input impedance (with and without load), (b) impedance of the stapes and cochlea, and (c) middle-ear transmission. With our measurements of changes in middle-ear transmission (for acoustic inputs ranging from 0.1 to 10 kHz) generated by stapedius contractions, this model is used to compute the changes in stapes impedance necessary to produce the measured changes in transmission (up to 30 dB). Results indicate the following. (1) The altered stapes impedance is representable by passive elements as its real part is positive. (2) As the magnitude of the stapes impedance increases with stronger muscle contractions both real and imaginary parts of the impedance change. (3) Changes in the stapes admittance are approximately proportional to changes in the transmission. (4) Changes in the middle-ear input impedance predicted by the model are compatible with measurements. Thus, the results support the hypothesis. [Work supported by NIH.]

10:15

**F6. Interactions among otoacoustic emissions, and associated threshold microstructures.** Arnold Tubis, Kenneth Jones (Department of Physics, Purdue University, West Lafayette, IN 47907), and Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

Third-order intermodulation distortion products or cubic difference tones (CDTs) in the ear canal may be generated from nonlinear cochlear interactions between two external tones, an external tone and a spontaneous otoacoustic emission (SOAE), or two (primary) SOAEs. In the last case [E. M. Burns *et al.*, *Hearing Res.* 16, 271–278 (1984)], the frequencies of the CDTs are found to be close to, but not always coincident with, frequencies of minima in the threshold microstructure, in contrast to the frequencies of the primary SOAEs. We have investigated this situation further by: (1) entraining one of the primary SOAEs with a variable frequency ( $f$ ) tone and (2) determining  $f$  for which the ear canal CDT level relative to that of the entraining tone is maximal. We generally find this value of  $f$  to correlate well with a threshold fine structure minimum. These results, along with related cochlear model simulations, are consistent with previous observations that CDT SOAEs tend to occur if the CDT frequency is sufficiently close to that of an evoked or spontaneous otoacoustic emission. [Work supported by NINCDS.]

10:30

**F7. Correlated changes in otoacoustic emissions and threshold microstructure during aspirin consumption.** Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907), Arnold Tubis, and Kenneth Jones (Department of Physics, Purdue University, West Lafayette, IN 47907)

The discovery that aspirin consumption can abolish spontaneous otoacoustic emissions [D. McFadden and H. S. Plattsmeyer, *J. Acoust. Soc. Am.* 76, 443–448 (1984)] provides a technique for further exploring the relation between otoacoustic emissions (spontaneous and evoked) and psychoacoustic threshold microstructure. Spontaneous emissions, delayed evoked emissions, synchronous evoked emissions, and threshold

microstructure were monitored before, during, and after consumption of 3.9 g of aspirin per day (three 325-mg tablets every 6 h). The changes in spontaneous emissions replicate the findings of McFadden and Plattsmeyer. Evoked emissions and threshold microstructure were also reduced by aspirin consumption but persisted longer and recovered sooner. In most instances the initial change in threshold microstructure was a reduction of threshold maxima with threshold minima remaining relatively constant. Further aspirin consumption elevated all thresholds to a level slightly above threshold maxima in the pre-aspirin measures. [Work supported by NIH.]

10:45

**F8. Sensitivity to pressure in the cochlea and the electromodel.** George C. Offutt (GoLo Center of Sensory Processes, Shepherdstown, WV 25443)

Stimuli were presented simultaneously to the chinchilla cochlea through the external meatus and the scala tympani so that displacements of the basilar membrane were controlled by dual pressure waves. Changes in the phase relationships of these pressure waves controlled the CM amplitude but often had little influence on the compound action potential. The  $N_1$  amplitude was apparently a function of the summed energy and was often independent of the phase difference between the two pressure waves. Thus, the mechanical displacements of the basilar membrane may not be the only source of energy for hair cell transduction in the cochlea. This requires a rethinking of the entire field of auditory research. It is possible that interpretations of all auditory research have been based upon a false assumption, and the inner hair cells (IHC) are primarily sensitive to electrical potentials. The tectorial membrane may be piezoelectric and perform the initial transduction of energy in the pressure wave into electrical potentials that are detected by the IHC. There is biological basis for proposing each of these transduction steps. The concept of electrosensitivity by the IHC provides the basis for a new view of the auditory system as presented in a recently published book, *The Electromodel of the Auditory System*.

11:00

**F9. Using ultrasonics to measure vibrational amplitudes of auditory organs in fish.** Mardi Cox and Peter H. Rogers (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A new underwater, noninvasive technique using ultrasonics has been devised to measure the amplitude response of the fish swim bladder and otolithic organs to a low-frequency sound wave. The body of the fish is scanned with a 10-MHz source while being subjected to a single frequency sound field. The resulting echo due to the impedance mismatch at the swim bladder and otoliths is received by another 10-MHz transducer, and then fed into a spectrum analyzer. The relative magnitude between the 10-MHz peak and the side bands created by the low-frequency excitation provides the desired amplitude information. Preliminary measurements have been obtained to demonstrate the method. Further refinements will allow measurements of  $25\text{ \AA}$  over 0.1 mm. This technique is superior to previous methods using accelerometers, microphones, Laser light scattering, and holographic interferometry because all of these require that the organs be either directly exposed to the measuring device or removed from the body. [Work supported by ONR.]

11:15

**F10. Model for the peripheral processing of sound in bony fish.** Peter H. Rogers, Mardi Cox (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332), Arthur N. Popper, and William M. Sidel (Georgetown University, School of Medicine-School of Dentistry, Department of Anatomy, Washington, DC 20007)

A new hypothesis for the processing of acoustical information by bony fish is presented. The hypothesis proposes that the ear itself can perform most of the calculations required to localize a sound source, and discriminate frequency and beam form for signal to noise enhancement. Prior theories assume all of the analysis occurs in the central nervous system

(CNS). A mechanism is demonstrated by which the ears may resolve the particle velocity into three vectorial components and determine the time derivative of the acoustic pressure. This information can be used to localize a sound source, assuming that the CNS is capable of evaluating the ratio of the appropriate velocity components. If localization is accomplished in this manner, it follows that the CNS would also be capable of finding the ratio between the magnitude of the acoustic particle velocity

and the time derivative of the acoustic pressure. This ratio is shown to be proportional to frequency in the farfield, and the product of the frequency squared and the distance from the source in the nearfield. Thus discrimination between frequencies is possible at any distance, absolute frequency can be determined in the farfield, and some capability to discriminate range might also be possible. Supporting evidence from the literature is presented. [Work supported in part by ONR and NIH.]

**TUESDAY AFTERNOON, 5 NOVEMBER 1985**

**ANDREW JACKSON HALL TENNESSEE PERFORMING ARTS CENTER, 1:30 TO 5:00 P.M.**

**Session G. Architectural Acoustics II, Engineering Acoustics I, and Musical Acoustics I: Modern Audio Recording Technology: Analog and Digital**

**Ludwig W. Sepmeyer, Chairman**

*Consulting Engineer, 1862 Comstock Avenue, Los Angeles, California 90025*

**Chairman's Introduction—1:30**

***Invited Papers***

**1:40**

**G1. Magnetic media, their characteristics and potential for sound recording and reproduction.** Marvin Camras (IIT Research Institute, 710 W. 35 Street, Chicago, IL 60616)

The physical characteristics of the available magnetic media which recommend them for audio sound recording and reproduction, by both analog and digital processes, will be discussed. Inherent characteristics of each medium which affect sound quality, dynamic range, and signal-to-noise ratio will be described.

**2:20**

**G2. Optical media, their characteristics and potential for sound recording and reproduction.** Ronald Uhlig and Allan Marchant (Eastman Kodak Company, Rochester, NY 14650)

An overview of the present-day optical processes, including film photography, magneto-optics, and laser-optics, will be discussed. The features of the processes and local defects of the media which affect sound quality, dynamic range, and signal-to-noise ratio will be described.

**3:00**

**G3. Error correction and concealment in digital recording.** Bart N. Locanthi (Pioneer North America, Inc., Development Laboratory, Duarte, CA 91010)

No medium is perfect. The imperfections range from accumulations of dirt on a film or disk to the microscopic absence of the recording medium on its supporting substrate. In digital reproduction, bits, or even words, are lost with potentially spectacular results. The nature of these defects, and the technologies that have been developed to overcome them, will be discussed and demonstrated.

**3:30**

**G4. Technical-esthetic considerations for sound recording.** John M. Eargle (JBL Inc., 8500 Balboa Boulevard, Northridge, CA 91329 and JME Associates, Los Angeles, CA 90068)

From the very beginning of sound recording, physical limitations of the medium have dictated certain esthetic constraints in the use of the medium. The major technical epochs have offered advantages to both engineer and producer, and the historical progression has provided solutions to problems in playing time, dynamic range, distortion, and space perspective. The problems associated with major format changes are almost predictable at this point. With each major change, producers and engineers have had to redefine the best fit of the message into the medium. They have invariably stumbled at first, and then rapidly learned to work the medium to the advantage of the music. In this paper we will review the fascinating interplay which has taken place between esthetics and technology over the years, concentrating on those problems associated with the coming of digital recording.

4:00

**G5. State of the art commercial music recording practice.** Richard P. Blinn (Capitol/EMI Corporation, Hollywood, CA 90028)

The variety of steps which take place between the artist's performance and the production of the final product for sale to the consumer will be explained and their effect on the final reproduced sound will be demonstrated by means of specially prepared recordings.

4:30-5:00

### **Panel Discussion**

George L. Augspurger, Moderator  
*Perception, Inc., Box 39536, Los Angeles, California 90039*

Panel discussion, question and answer period, and more demonstration recordings.

**PANEL MEMBERS:** Marvin Camras  
Ronald Uhlig  
Allan Marchant  
Bart N. Locanthi  
John M. Eargle  
Richard P. Blinn

**TUESDAY AFTERNOON, 5 NOVEMBER 1985**

**DAVIDSON ROOM A, 1:30 TO 5:00 P.M.**

### **Session H. Biological Response to Vibration I: Tactile Aids for the Hearing Handicapped**

Ronald T. Verrillo, Chairman  
*Institute for Sensory Research, Syracuse University, Syracuse, New York 13210*

#### ***Invited Papers***

1:30

**H1. Tactile devices in sensory substitution systems.** Carl E. Sherrick (Department of Psychology, Princeton University, Princeton, NJ 08544)

An overview of the research on tactile aids for the deaf, blind, and deaf-blind will pay particular attention to the problems of encoding the environmental signals in ways that the skin senses can interpret them efficiently. Following a brief summary of the early approaches to the development of systems for sensory substitution, there will be a closer examination of the efforts mounted in the past 20 years. The talk is intended to include illustrations of successful and unsuccessful efforts at device and system development from scientific and engineering studies. In addition, there will be recounted entrepreneurial experiences of a number of investigators and developers of devices and methods for improving communicative skills in the sensorily disabled population. [Work supported by NIH Grant 04755.]

**H2. The application of principal-components spectral analysis to sensory aids for the hearing impaired.** Stephen A. Zahorian (Department of Electrical Engineering, Old Dominion University, Norfolk, VA 23508)

In order to present speech information through an alternate sensory modality—such as touch or vision—extensive processing of the speech signal is required. This processing should represent a large amount of speech information with a small number of slowly varying parameters. These parameters should be relatively speaker independent, robust in the presence of noise, and closely linked to the perceptually important features of speech spectra. One such set of parameters are speech spectral principal components. In this paper we will discuss the application of principal-components analysis of speech spectra to speech recoding for use with sensory aids. A brief overview of the analysis procedure will be included. An interpretation of principal components, and a comparison with other speech parameters such as speech formants, will be presented. Both potential benefits and drawbacks of this analysis procedure will be discussed. In addition, the implementation of a vowel articulation training aid based on principal components will be presented. Although the vowel training aid utilizes a visual display format, with color as a primary information-bearing parameter in the display, the procedure used for obtaining principal components applies equally well to a tactile aid.

2:30

**H3. Tactile aids: A comparison of single and multichannel devices.** Arlene Earley Carney (Department of Speech and Hearing Science, University of Illinois, Champaign, IL 61820)

The use of tactile aids as sensory prosthetic devices for the deaf has become a topic for discussion among researchers and clinicians interested in the hearing impaired. The importance of work in this area has become even greater with the increasing frequency of cochlear implants in deaf adults and children. A primary question in the area of tactile research concerns the design of the tactile device which will provide the most appropriate stimulus to the deaf subject. This presentation will focus on several studies which compared the use of a multichannel tactile device with a single-channel device for the perception of both segmental and suprasegmental aspects of speech. Results of these experiments indicated that the single-channel device provided more accurate cues for the perception of suprasegmentals, such as number of syllables per word, syllabic stress, and intonation. Further, there was no significant difference between single- and multichannel devices for the perception of vowels and consonants. An additional experiment was done with deaf patients who had received a single-channel cochlear implant, using the same segmental and suprasegmental stimuli as in the tactile studies. A comparison of these results to those obtained with a single-channel tactile device showed no advantage for the implant, despite the wearability and experience factors in its favor. These results will be discussed in view of the long-term goals for design and tactile aids, and for their most effective use as speech training aids for the deaf.

3:00

**H4. The case for tactile aids.** James D. Miller and Janet M. Weisenberger (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Devices that convert sound information into vibrotactile stimulation have been shown in laboratory and clinical studies to enable persons to appreciate many aspects of the acoustic environment, and thus are of potential benefit to deaf persons. These vibrotactile devices provide information comparable to that obtained with surgical cochlear implants, but appear to have some advantages, particularly in use with children. These advantages include no invasive surgery, the ability to accommodate growth of the user, the potential to use the device for a trial period to assess its usefulness, the possibility of shared use by more than one person, and considerably lower cost. At CID, a number of vibrotactile aids ranging from single-channel commercially available aids to multichannel tactile vocoders are being evaluated. Results from studies in our own and other laboratories will be used to address such issues as the features of the acoustic environment that can or should be transmitted through a tactile aid, the differences in the information transmitted by commercially available versus experimental laboratory devices, and the technical problems that remain to be overcome in the design and evaluation of improved vibrotactile aids. [Work supported by NIH and NSF.]

### *Contributed Papers*

3:30

**H5. A case study of the tracking skill of a deaf person with long-term tactile aid experience.** Roger W. Cholewiak and Carl E. Sherrick (Department of Psychology, Princeton University, Princeton, NJ 08544)

The paper describes a case study of Dimitry Kanievski, a deaf individual who has been using a vibrotactile aid for approximately 13 years. He has acquired the ability to lipread speakers in three languages, using the Kanievski speech-analyzing device. The report describes his communica-

tive abilities with and without the aid in his native language, Russian, as well as in English and Hebrew. When Dr. Kaniewski was tested with the De Filippo-Scott tracking technique, the aid provided for a considerable improvement in performance over unaided lipreading scores. The degree of improvement, however, was a function of several factors, in particular his unaided lipreading rate for the different languages. Plotting the ratio of aided to unaided performance against unaided tracking rate yielded a power function. [Work supported by NIH Grant NS-04775.]

3:45

**H6. Speech reception in deaf adults using vibrotactile aids or cochlear implants.** Paul Milner and Carole Flevaris-Phillips (Audiology and Speech Pathology Section, V. A. Medical Center, West Haven, CT 06516)

Many post-lingually deafened adults rely heavily upon speechreading as the primary source of speech information. Vibrotactile aids or cochlear implants may help augment the limited visual cues of speechreading, especially the ones which permit voiced/voiceless and nasality distinctions. Measures of the apprehension of voicing and nasality, as well as other attributes of speech reception, were obtained using the Diagnostic Rhyme Test [W. D. Voiers, in *Benchmark Papers in Acoustics*, Vol. 11, edited by M. Hawley, pp. 374-387 (1977)] with candidates for possible cochlear implantation. Results using vibrotactile aids or cochlear implants either individually or in combination with speechreading indicated that both vibrotactile aids and cochlear implants provided improvements in the perception of nasality, but voiced/voiceless distinctions were not as clearly recognized. [Work supported by Veterans Administration Rehabilitation Research and Development Service.]

4:00

**H7. Pitch detection on a programmable vibrotactile aid for the deaf.** Silvio P. Eberhardt (Sensory Aids Laboratory, Department EECS, Johns Hopkins University, Baltimore, MD 21218)

A vibrotactile aid capable of real-time speech processing has been developed. Three microprocessors are used for, respectively, speech sampling and data reduction, pattern matching, and generation of vibrator drive signals. Data reduction is performed by generating data records at each zero crossing of the speech signal. Records consist of the interval since the last zero crossing (ZCI) and average amplitude during that interval. A pitch detector has been implemented on the aid. The algorithm is based on the observation that consecutive pitch periods have similar ZCI and amplitude structures. Pattern matching is performed by comparing adjacent strings of ZCI and amplitude values. Sum-of-difference correlation values are low when the strings correspond to actual adjacent pitch periods. The pitch search window is adaptively narrowed after voicing has been detected. The detector shows no tendency to lock onto other harmonics or formants. Results of comparisons with other pitch detectors in several white and babble noise conditions will be presented. [Work supported by NIH, NSF, NIHR.]

4:15

**H8. Vibrotactile sensitivity thresholds of hearing children and of profoundly deaf children.** Lynne E. Bernstein, Miriam B. Schechter, and Moise H. Goldstein, Jr. (Sensory Aids Laboratory, Department of Electrical Engineering and Computer Science, Johns Hopkins University, Baltimore, MD 21218)

We investigated sensitivity thresholds for 1-s sinusoidal stimuli of 20, 40, 80, and 160 Hz with hearing 5- to 6- and 9- to 10-year-olds and adults. Stimuli were presented in a two-interval forced choice procedure according to an adaptive rule to estimate the 70.7% threshold [H. Levitt, *J. Acoust. Soc. Am.* 49, 467-477 (1971)]. Results showed that young chil-

dren were somewhat less sensitive than older children and adults. The present results are not in complete agreement with published reports [R. T. Verrillo, *Bull. Psychonomic Soc.* 9, 197-200 (1977) and R. D. Frisina and G. A. Gescheider, *Percept. Psychophys.* 22, 100-103 (1977)]. Previously, we found no effect of age with haversine pulse trains. To explore whether hearing status might affect sensitivity, two prelingually profoundly deaf children were tested with 1-s haversine stimuli at 10, 50, 100, 160, and 250 pulses per second. The deaf children were at least as sensitive as previously tested normally hearing adults and children. [Work supported by NSF and NINCDS.]

4:30

**H9. Effects of aging on vibrotactile temporal resolution.** Clayton L. Van Doren, Grace L. Lanni, Preeti Verghese, Ronald T. Verrillo, and George A. Gescheider (Institute for Sensory Research, Syracuse University, Syracuse, NY 13203)

Psychophysical thresholds for detecting a temporal gap centered in a background stimulus were measured by two-interval forced-choice tracking. The 16- to 256-ms gaps were flanked by 350-ms bursts of either 250-Hz sinusoidal vibration or bandpass noise. In each trial the threshold intensity was measured for detecting a gap of a fixed duration. This threshold decreased as gap duration increased and could be approximated by a power function of the form  $I = A(G/G_0)^{-B} + C$ . In this expression,  $I$  is the gap-detection threshold in dB SL,  $G$  is the gap duration in ms,  $G_0$  equals 1 ms,  $A$  and  $C$  are constants in dB SL, and  $B$  is a dimensionless constant.  $A$ ,  $B$ , and  $C$  increased with subject age. The primary effect of increasing age was a reduced temporal resolution for gaps smaller than 64 ms. This reduced sensitivity may effect tactile perception of some speech features such as voiced and unvoiced stop consonants in older subjects. [Work supported by NIH Grant NS-09940.]

4:45

**H10. Field tests of a wearable 16-channel electrotactile sensory aid in a classroom for the deaf.** Frank A. Saunders (Smith-Kettlewell Institute of Visual Sciences, 2232 Webster Street, San Francisco, CA 94904) and Barbara Franklin (Department of Special Education, San Francisco State University, San Francisco, CA 94132)

Field testing of a wearable electrotactile sensory aid began in February 1985 at the Jackson Hearing Center, an oral program for deaf children in Palo Alto, CA. Six children, 3 to 8 years of age, participated in the study. Each had a profound congenital bilateral sensorineural hearing loss in excess of 105 dB. The wearable sensory aid, the Tacticon model 1600, presents 16 channels of frequency information via a tactile belt worn around the abdomen. Wearing time for each child was gradually increased from half-hour daily lessons to 4 h of use per day. Environments included the classroom, outside recess with physical education activities, and field trips. Receptive training was directed at recognition of both suprasegmental and segmental features of speech. Suprasegmental features included duration, number of syllables, rhythm, and stress. Segmental features included the recognition of specific speech sounds in isolation, beginning with the Ling 5 sound test of /a/, /e/, /u/, /s/, and /sh/. The assessment procedure was conducted in three modes: (a) aided residual hearing alone, (b) aided hearing plus lipreading cues, and (c) aided hearing plus lipreading plus tactile cues from the sensory aid. The addition of tactile cues results in a significant increase in the discriminability of both suprasegmental and segmental features. The children also received experience with the sensory aid during speech training, attending to the tactile patterns resulting from different speech features such as nasality, voicing, friction, and plosion, comparing their utterances with those of the teacher. Their performance indicated that the tactile feedback supported an improvement in speech production, as well as reception. [Work supported by SBIR/NIH: 2 R44 AG/NS04817.]

## Session I. Psychological Acoustics I: Binaural Hearing

D. Wesley Grantham, Chairman

*Bill Wilkerson Hearing & Speech Center, 114 19th Avenue South, Nashville, Tennessee 37212**Contributed Papers*

1:30

**11. The effects of phase randomization in the detection of interaural differences of time in five-component complexes.** R. H. Dye, Jr. and W. A. Yost (Parham Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

Threshold interaural differences of time (ITDs) were measured for two types of interaural delays. In the first case, the stimulus consisted of 550–650–750–850–950 Hz, and thresholds were measured for all possible combinations of one, two, three, four, and five delayed components. In the second case, threshold interaural envelope delays were measured for five-component waveforms whose carriers were 2 and 4 kHz and whose modulation frequencies ranged from 20–500 Hz. Comparisons were made between thresholds measured when all components were added in sine phase and those obtained when the starting phases were randomized between intervals of a two-interval task. The signals were 100 ms in duration, and the level of each component was 65 dB SPL. For the low-frequency waveforms in which a subset ( $m$ ) of the five components was delayed, phase randomization had a negligible effect on performance. Regardless of the starting phases,  $d'_m$  was approximately equal to  $d'_1$ , where  $d'_1$  is the  $d'$  when one of the five components was delayed. These data tend to support the contention that this task is performed by a component-by-component analysis, with the lateral position determined by a weighted average of the information at each frequency. For the high-frequency waveforms with envelope delays, phase randomization had only small deleterious effects on performance, rendering untenable explanations based on interaural comparisons of the outputs of simple envelope detectors. [Work supported by NSF, NIH, and AFOSR.]

1:45

**12. Spectral dominance in sensitivity to interaural delay for broadband stimuli.** P. M. Zurek (Research Laboratory of Electronics, Room 36-736, Massachusetts Institute of Technology, Cambridge, MA 02139)

Just-noticeable differences (jnds) in interaural delay of a target band were measured in the presence of a diotic spectral fringe. Stimuli were clicks or 100-ms bursts of noise with flat spectra from 0.1 to 9 kHz. The bandwidth of the target band was 0.4 times the center frequency. With either clicks or noise, the jnds for target bands having center frequencies between 0.5 and 1 kHz were roughly unaffected by the presence of the fringe: They were on the order of tens of microseconds. In contrast, the jnds for target bands at high frequencies were greatly degraded by the presence of the fringe: They increased from approximately 100  $\mu$ s to greater than 600  $\mu$ s. With noise, delaying a low-frequency target band causes the image of that band to separate from the image of the fringe. With clicks, delaying a low-frequency band (with about 3% of the total energy) causes the entire wideband image to lateralize. In general, the results suggest that the degree of the dominance region described by Bilsen and Raatgever [Acustica 28, 131–132 (1973)] is virtually total and that interaural delays at high frequencies are insignificant compared to those at low frequencies. [Work supported by NIH.]

2:00

**13. Perception of modulations in pitch and lateralization.** Laurel Beecher and Richard M. Stern (Department of Electrical and Computer Engineering and Bioengineering Program, Carnegie-Mellon University, Pittsburgh, PA 15213)

We compare the ability to perceive low-frequency sinusoidal modulations of monaurally and dichotically created pitch to the perception of modulations of the subjective lateral position of a binaural image. The stimuli used in the dichotic pitch experiments had a 0- to 2000-Hz lowpass spectrum, and they were created by modulating the time delay of a multiple-phase-shift filter [F. A. Bilsen, J. Acoust. Soc. Am. 59, 467–468 (1976)]. Stimuli with similar spectra and sinusoidally modulated interaural time delays (ITDs) were used for the lateralization experiments [D. W. Grantham and F. L. Wightman, J. Acoust. Soc. Am. 63, 511–523 (1978)]. We also examined the perception of monaural frequency-modulated pure tones, and monaural lowpass stimuli created by summing the signals to the two ears from the dichotic pitch stimuli. Subjects discriminated between sinusoidally modulated and unmodulated stimuli using 2IFC paradigms, and we determined the threshold frequency deviation or ITD as a function of modulation frequency. Preliminary results were similar for all three pitch experiments: Frequency deviation at threshold was constant for modulation frequencies up to approximately 10 Hz, and then increased as a power function of modulation frequency up to at least 32 Hz. Threshold ITDs in the lateralization experiments were constant for frequencies up to about 4 Hz, and then increased with increasing modulation frequency. [Work supported by NIH.]

2:15

**14. Lateralization of dichotic noise bursts: Trading onset against offset disparity.** Arnold M. Small and Francis Kuk (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

We previously reported on the lateralization of a sound image associated with either onset or offset disparity of dichotically presented uncorrelated noise bursts [J. Acoust. Soc. Am. Suppl. 1 70, S91 (1984)]. We found that (1) onset and offset disparities are both effective cues, with onset being more potent, (2) lateralization is toward the leading ear for onset and toward the lagging ear for offset disparity, (3) shorter stimuli exhibit stronger lateralization effects, and (4) equalization of loudness of the dichotic stimuli does not alter lateralization (thus ruling out simple loudness as the cue mediating the effect). The present research investigated the interaction of onset and offset disparity by covarying each. Thus the dichotic noise bursts could differ in onset or offset disparity or both, with either ear leading. Listeners quantified image location by positioning a slider in a diagram of a head. The results indicated that a trading relation existed. That is, a given image position could be achieved by a variety of simultaneously present onset and offset disparities. Onset disparity was by far the stronger influence.

2:30

**15. Signal onset times and the precedence effect.** Brad Rakerd (Department of Audiology and Speech Sciences, Michigan State University, East Lansing, MI 48824) and W. M. Hartmann (Physics Department, Michigan State University, East Lansing, MI 48824)

A consistent finding in our studies of auditory localization in rooms has been that brief tones with abrupt onsets and offsets are localized far more accurately than are sustained tones with slow onsets [W. M. Hartmann, J. Acoust. Soc. Am. 74, 1380–1391 (1983); B. Rakerd and W. M. Hartmann, J. Acoust. Soc. Am. Suppl. 1 75, S88 (1984)]. The purpose of the present study was to explore the contribution of signal onset rate to this effect. In a room with left-wall acoustical reflections of brief delay (mean delay time = 1.5 ms), we assessed localization performance for sus-

tained tones (500 Hz, 60 dB) with onsets of 5000, 1000, 500, 100, 50, 10, 5, and 0 ms. The accuracy of subjects' ( $n = 4$ ) localization judgments increased monotonically with rate for all onsets less than 100 ms. Testing with a subset of the onset times showed that a similar pattern of results is obtained in rooms with other paths of reflection (ceiling, mean delay = 1.5 ms) and with other delay times (left wall, mean delay = 9 ms). We hypothesize that signal onsets must be shorter than 100 ms if they are to trigger operation of the precedence effect in rooms, and that within that regime their effectiveness increases with increasing rate. [Work supported by NIMH.]

2:45

**I6. Psychoacoustic results supporting the assumption of binaural inhibition mechanisms.** Werner Lindemann (Lehrstuhl für allgemeine Elektrotechnik und Akustik, Ruhr-Universität, D-4630 Bochum, Postfach 102148, West Germany)

This paper describes localization and lateralization experiments with stationary and nonstationary signals, the results of which can be explained by inhibitory mechanisms in binaural signal processing. Multiple auditory events are discussed which can be perceived with pure-tone stimuli at various combinations of interaural time and level differences. For noise signals, multiple auditory events are reported which occur as a function of the degree of interaural coherence. Inhibitory mechanisms as related to the law of the first wave front are investigated with pairs of impulses presented dichotically via headphones. The psychoacoustic results can be predicted by simulations based on a binaural cross-correlation model—extended by inhibitory mechanisms [W. Lindemann, *J. Acoust. Soc. Am. Suppl. 1* 74, S85 (1983)]. [Work supported by the Deutsche Forschungsgemeinschaft.]

3:00

**I7. The relationship between the cross-correlation function of two-channel acoustic signals reproduced through the headphones and spatial**

**impressions of sound image.** Kohichi Kurozumi and Kengo Ohgushi (NHK Science and Technical Research Laboratories, 1-10-11 Kinuta, Setagaya, Tokyo 157, Japan)

White noises with various cross-correlation functions are reproduced through headphones. Subjects are asked to make similarity judgments and some subjective evaluations of pairs of noises. The experimental data are analyzed by Kruskal's multidimensional scaling (MDS) programs. The analysis of the experimental data shows the following: (1) spatial impressions of sound image depend mostly on the width, the elevation, and the lateral displacement of the sound image; (2) the width of the sound image increases as the maximum absolute value of the cross-correlation function decreases, and as its peak position goes away from  $\tau = 0$ ; (3) the elevation of the sound image tends to increase as the maximum value of the cross correlation function increases, as long as the maximum value is positive, (4) the lateral displacement depends on the asymmetry of the shape of the cross-correlation function and mostly on the positive peak position of it.

3:15

**I8. Masking level difference in infants.** Robert J. Nozza (Division of Audiology, Children's Hospital of Pittsburgh, Pittsburgh, PA 15213)

The binaural masking level difference (MLD) for a 500-Hz pure tone was estimated for infants between 6 and 8 months of age. A masker with bandwidth of 600 Hz (No = 35 dB SPL), geometrically centered at the test frequency, was presented in phase to a pair of TDH-49 earphones. Each subject was tested in each condition (NoSo and NoSn) in a single session, the sequence counterbalanced among the subjects. A computer-controlled visual reinforcement head-turn procedure was used. Masked threshold was estimated using a simple up-down staircase procedure with a 5-dB stepsize. Infant MLD estimates compare favorably with those of an adult control group. However, the infants' masked-threshold estimates, used in computing the MLD, were evaluated relative to the adult group for each condition. [Work supported by Deafness Research Foundation.]

TUESDAY AFTERNOON, 5 NOVEMBER 1985

REGENCY BALLROOM IV, 1:30 TO 4:30 P.M.

## Session J. Speech Communication I: Suprasegmentals: Perception and Production

Robert F. Port, Chairman

*Department of Linguistics, Indiana University, Bloomington, Indiana 47405*

### Contributed Papers

1:30

**J1. Stress groups and rhythm in American Spanish.** Guillermo A. Toledo (Laboratorio de Investigaciones Sensoriales, CONICET, CC53, 1453, Buenos Aires, Argentina)

This paper reports a description of rhythm in several discourses both in spontaneous speech and in the oral reading of narrative prose, essay, poetry in free verse, and sonnets in American Spanish. These speech materials were digitized and an acoustic study of the duration of stress groups—a word with primary stress plus unstressed words in proclitic and enclitic positions—was undertaken. A variance analysis of percentage deviations of these stress groups showed values in an ample range: a lower degree of variability in poetry reading and a higher degree in prose reading and spontaneous speech. Also, there was a certain tendency to temporal equalization in stress groups of different sizes and, inversely, a tendency to proportionality between the increment in size and the increased duration. This would appear to suggest that each discourse would be rhythmically supported by a twofold tendency, syllable- and stress-timing, combined in a free scheme. This claim agrees with previous re-

search providing evidence on interstress interval patterns of similar oral discourses in American Spanish [G. Toledo, *J. Acoust. Soc. Am. Suppl. 1* 77, S53 (1985)].

1:45

**J2. Fundamental frequency and its relationship to Spanish stress.** M. I. Massone and A. M. B. de Manrique (Laboratorio de Fonética Experimental, Universidad Católica Argentina, Bartolomé Mitre 1869, 1039 Buenos Aires, Argentina)

In the present work, we attempt to describe  $F_0$  movements and the amount of  $F_0$  change in the stressed syllables of Spanish sentences as a function of phonetic and linguistic effects. The material consisted of a list of 92 sentences with different syntactic structures which were recorded twice by a male speaker. Wideband spectrograms and  $F_0$  contours of all the sentences were obtained.  $F_0$  movements on the syllabic nucleus of each accented syllable were chosen as the parameters for the analysis of  $F_0$  contours. Results showed that the accented syllabic nucleus presents the

following *F0* movements: rise, fall, level, and rise-fall. These movements are correlated with effects such as the nature of the preceding consonant (level with voiceless, rise with voiced), position of the accented word (fall in the last accented word, rise in yes-no questions), and boundary (rise-fall or fall). Emphatic stress is correlated with increased *F0* and larger *F0* excursions. The amount of *F0* change seems to follow a declination line. This general pattern is altered by different effects: lexical (adverbs, pronouns), syntactic (phrasal verbs, apposition, prepositional phrases, clauses, syntactic phrases), and semantic (modality operators, emphasis, prominent words).

2:00

**J3. The effect of vowel quality variations on stress-beat location.** Robert Allen Fox (Speech & Hearing Science, The Ohio State University, 324 Derby Hall, 154 N. Oval Mall, Columbus, OH 43210) and Ilse Lehiste (Department of Linguistics, The Ohio State University, 1841 Millikin Road, Columbus, OH 43210)

This study examines the effect which vowel quality variations have upon the location of vowel onsets—indicators of stress-beat location—in rhythmically produced lists of monosyllables. Three talkers produced 120 sequences consisting of seven monosyllabic tokens of the form /s\_\_t/. These token sequences were either homogeneous (identical components) or alternating (stimulus syllables alternated with the token "sight"). Talkers produced the sequences in both a metronome and nonmetronome condition. Analysis of the obtained results shows that vowel quality variations had a significant effect upon the location of the vowel onset. Regression analysis demonstrated that the differences in vowel onset-to-vowel onset intervals were significantly related to vowel duration differences between the tokens. There was also a significant difference between talkers in terms of the location of the vowel onsets, suggesting either the use of different production strategies or differential capabilities in doing the rhythmic task. No evidence of preboundary lengthening was obtained. These data support the suggestions [R. A. Fox and I. Lehiste, *J. Acoust. Soc. Am. Suppl.* 1 77, S54 (1985)] that the location of the "stress beat" is determined on the basis of the acoustic structure of the entire syllable, rather than strictly upon the articulatory onset of the vowel and that subjects in such a task do not necessarily impose a hierarchical structure on the syllabic sequences. Perceptual data relevant to the importance of vowel durations to the stress-beat location will also be presented.

2:15

**J4. The role of duration in the perception of consonants within reduced syllables.** Danièle Archambault (Université de Montréal and INRS—Télécommunications, 3 Place du Commerce, Nuns' Island, Verdun, Québec, Canada H3E 1H6)

The role of duration as a perceptual cue to the presence of two underlying consonants versus one is examined in Montreal French. Due to deletion of the intervening phonetic segments, two identical syllable-initial consonants may be joined in one articulation such that, for example, two plosives show a single burst. A series of utterances in which these double consonants and the corresponding single ones appear in identical segmental contexts was recorded and then presented to listeners in perceptual tests [e.g., *la plupart* (most of it) → /laplypar/ → [lap:ar] as opposed to *la part* (the part) → /lapar/ → [lapar]]. To insure that listeners base their judgment on duration to distinguish between the utterances, they were also presented with the same utterances in which, by use of signal processing, the durations of the consonants were inverted, i.e., the double consonant was given the relative duration of the single consonant and vice versa. Results show that while a long duration consistently leads listeners to recognize a double consonant, a short duration does not automatically prompt them to identify a single consonant. Results also indicate that consonant duration must be considered in a larger context than that of the syllable. [Research done at the Université de Montréal and supported by the National Research Council of Canada and Québec Government.]

2:30

**J5. Perceived pitch (tonality) of five unfiltered nonsense syllables and their identification and quality assessment through six 1-oct filter bands.**

Carole E. Johnson and Carl W. Asp (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740)

Research involving the perceived pitch (tonality) [M. M. Peterson and C. W. Asp, *J. Acoust. Soc. Am. Suppl.* 1 52, S146 (1972) and C. W. Asp, J. S. Berry, and C. S. Bessell, *J. Acoust. Soc. Am. Suppl.* 1 64, S20 (1978)] and optimal octaves in phoneme perception [E. E. McKenney and C. W. Asp, *J. Acoust. Soc. Am. Suppl.* 1 51, S122 (1972) and R. M. Miner and J. L. Danhauer, *J. Am. Audiol. Soc.* 2, 163-168 (1977)] has supported the notion that phonemes may be "frequency specific." The purposes of this study were to have normal-hearing listeners (1) rate the perceived pitch of five unfiltered nonsense syllables with phonemes homogeneously grouped according to a pitch model [C. W. Asp, J. S. Berry, and C. S. Bessell, *J. Acoust. Soc. Am. Suppl.* 1 64, S20 (1978)]; and (2) identify and assess the quality of the nonsense syllables each filtered through six 1-oct filter bands with center frequencies of 250, 500, 1000, 2000, 4000, and 8000 Hz. Results indicated that (1) the nonsense syllables could be ranked according to low to high perceived pitch (i.e., /mumu, vovo, lala, keke, and sisi/); (2) identification scores for the filtered nonsense syllables were high except when filtered through the two lowest bands (i.e., 250 and 500 Hz); and (3) quality assessment of the filtered nonsense syllables revealed "optimal octaves" for perception of each nonsense syllable. Results are discussed in terms of the rankings of perceived pitch of the unfiltered nonsense syllables in relation to their filtered "optimal octaves." Implications for aural rehabilitation will be discussed.

2:45

**J6. Perceived pitch (tonality) of nonsense syllables with phonemes homogeneously or heterogeneously grouped phonemes according to a pitch model.** Carl W. Asp and Carole E. Johnson (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740)

Research involving the perceived pitch (tonality) [M. M. Peterson and C. W. Asp, *J. Acoust. Soc. Am. Suppl.* 1 52, S146 (1972) and C. W. Asp, J. S. Berry, and C. S. Bessell, *J. Acoust. Soc. Am. Suppl.* 1 64, S20 (1978)] and optimal octaves in phoneme perception [E. E. McKenney and C. W. Asp, *J. Acoust. Soc. Am. Suppl.* 1 51, S122 (1972) and R. M. Miner and J. L. Danhauer, *J. Am. Audiol. Soc.* 2, 163-168 (1977)] has supported the notion that phonemes may be "frequency specific." For these studies, homogeneously grouped phonemes were selected according to a pitch model [C. W. Asp, J. S. Berry, and C. S. Bessell, *J. Acoust. Soc. Am. Suppl.* 1 64, S20 (1978)]. The purpose of this study was to determine the effects of homogeneously and heterogeneously grouping phonemes according to the pitch model. This included pairing a low /m/, a middle /l/, and a high /s/ consonant each with a low /u/, a middle /a/, and a high /i/ vowel. This resulted in three homogeneous (i.e., /mu, la, and si/) and seven heterogeneous (i.e., /ma, mi, lu, li, su, and sa/) nonsense syllables. Ten normal-hearing young adult listeners selected the syllable with the highest pitch in each pair within a paired-comparison paradigm. Results indicated that (1) the nonsense syllables could be ranked from high to low perceived pitch (i.e., /si, li, sa, mi, la, su, ma, lu, and mu/); (2) both consonants and vowels affected the listeners' judgments, although vowels appeared to have a greater effect; and (3) the rank order of the perceived pitch of the nonsense syllables was in agreement with past research. Tonality is discussed in terms of acoustic data as well as a mathematical perceptual model for predicting the pitch perception of phonemes in syllables and in words.

3:00

**J7. Perceptual cues to lexical accent contrasts in English and Japanese.** Mary E. Beckman (Linguistics and Artificial Intelligence Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

Earlier experiments have shown that Japanese and English differ in the extent to which they use duration and vowel amplitude patterns as acoustic correlates of lexical accent [M. E. Beckman, *J. Acoust. Soc. Am. Suppl.* 1 71, S23 (1982); *J. Acoust. Soc. Am. Suppl.* 1 75, S41 (1984)]. A new experiment examines the relative salience of various acoustic correlates as perceptual cues in the two languages. Minimally contrasting test utterance pairs from the earlier production experiments were used to synthesize test stimuli with all possible combinations of acoustic patterns (e.g., the *F0* and amplitude patterns of *contráct* combined with the dura-



tion and spectral quality patterns of *contract*). Japanese, Americans with speaking knowledge of Japanese, and monolingual Americans judged the accent patterns of these test stimuli. Results showed differences correlated both with the source language of the stimulus, and with the language background of the subject. For all three groups, *F0* was the dominant cue to the accent patterns of the Japanese stimuli, scoring significantly higher than it did in the judgments of the English stress patterns by the same group. In judging the English stimuli, the monolingual American subjects relied on acoustic cues other than *F0* to a greater extent than did the bilingual Americans, although both American groups used them to a greater extent than did the Japanese subjects, most of whose scores for amplitude, duration, and spectral quality patterns were at chance.

3:15

**J8. Limits of temporal discrimination for sequences of speech sounds.** Daniel A. Gardner, Brad S. Brubaker, and Richard M. Warren (Department of Psychology, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

Earlier studies from a number of laboratories found that the order of vowels in iterated sequences of three or four items could not be identified below durations of about 100 ms/item. The present study required untrained subjects to distinguish between iterated sequences of three vowels (/i/, /a/, /u/) presented in the same or in permuted order. As would be anticipated, distinguishing between permuted orders could be accomplished through naming of individual items at durations greater than 100 ms, but contrary to some theories, metathesis resulting in perceptual equivalency of different orders did not occur at item durations below 100 ms. Rather, permuted orders were readily distinguishable down to durations corresponding to single glottal pulses, although component vowels could not be recognized. These results are consistent with theories considering that comprehension of speech does not require the ability to detect component speech sounds and their orders, but is based instead on recognition of unresolved temporal patterns. [Work supported by NIH and NSF.]

3:30

**J9. The effect of sentence timing on the perception of word-initial stop consonants.** Gary R. Kidd (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

The influence of temporal context on the perception of voice onset time (VOT) as information for the identity of a word-initial stop consonant was examined in three experiments. Different versions of a ten-word precursor phrase were constructed by recording the phrase at a fast and a slow rate and then combining words from the two original phrases to produce composite phrases with various patterns of rate changes. A final (target) syllable from a seven member /gi/-/ki/ VOT continuum constructed from natural speech was added to the end of the phrase after a variable pause duration. Subjects listened to each phrase version and judged the identity of the target syllable. In general, the VOT boundary was found to shift to shorter values with precursors that contained more fast words and shorter pause durations. Furthermore, while the rate of the words immediately preceding the target appeared to have the greatest effect, there were also significant rate effects due to the rate of stressed words early in the precursor and the pattern of rate changes throughout the precursor. [Work supported by NIMH.]

3:45

**J10. Use of timing to discriminate words.** William Reilly and Robert Port (Department of Linguistics, Indiana University, Bloomington, IN 47405)

We sought to use timing measures alone to discriminate words with similar sequences of basic acoustic segments. Previous work by Waibel [J. Acoust. Soc. Am. Suppl. 1 72, S32 (1982)] and others showed that a crude segmental classification scheme could reduce the search space of the English lexicon to cohorts of fairly small size. We took two sets of six similar words, differing primarily in stress, voicing, and vowel length, including *bacon, begin, Begin, pecan* and *zipper, supper, suburb*. Thus they might fall in similar cohorts with a gross segmental classification. The words were spoken twice in a carrier sentence at two tempos by 12 speakers. Durations of segmental intervals (vowel and obstruent durations, VOT, etc.) were measured from sound spectrograms. Discriminant analysis was run on the set of first tokens and the derived classificatory equations were used to predict word identity for both the first and second set of tokens. Prediction accuracy for both sets was about 86% where chance performance is about 8%. That is, timing information about the whole word permitted very good identification from among cohorts of segmentally similar words. These results suggest that appropriate timing measures could be effective discriminators in whole-word speech recognition.

4:00

**J11. How cryptic is distorted speech?** Maria Wingstedt and Richard Schulman (Department of Linguistics, Stockholm University, 10691 Stockholm, Sweden)

A series of tests was conducted examining the ability of native Swedish speakers to learn to comprehend phonologically distorted speech. Two versions of the same text were produced: one with a heavy Chinese accent and one with an artificially produced accent (Cryptic) which was very different in character from the Chinese one. Two groups of 22 subjects listened to either one of the versions. A list of isolated words produced only with the Cryptic accent followed both of the text versions. A third group, acting as a control, listened only to the isolated Cryptic word list. It was shown that the group exposed to the Cryptic text version had almost double the comprehension of the word list as those exposed to either the Chinese accent or to no text prior to the list—both performing equally poorly. We conclude that learning to compensate for a particular accent is specific to that accent and its comprehension will not be augmented by prior exposure to a very different accent, and that this result should hold true for distortions in general. [Work supported by HSFR.]

4:15

**J12. Perception of stress pattern and word recognition.** Vincent J. van Heuven (Department of Linguistics/Phonetics Laboratory, Leyden University, P.O. Box 9515, 2300 RA Leiden, The Netherlands)

Theories of spoken word recognition do not explicitly take account of nonsegmental properties of the stimulus. The present research investigates how, and to what extent, acoustic information on the stress pattern of a (Dutch) word contributes to word recognition, both in hifi degraded natural speech and in synthetic speech (concatenated diphones). In a series of experiments, words with correct or incorrect stress position in the first, second, or third syllable were presented in isolation or preceded by a short carrier phrase. Words were made audible in fragments of increasing size ("gating") so as to trace the respective contribution of stressed and unstressed syllables to the word isolation process as the stimulus develops in time. Whole word recognition was included in some of the experiments as a control task. Selected results will be presented and some of the consequences for word recognition models will be discussed.

**Session K. Underwater Acoustics II: Propagation**

Stephen N. Wolf, Chairman  
*Naval Research Laboratory, Washington, DC 20375*

Chairman's Introduction—1:30

**Contributed Papers**

1:35

**K1. Causality, caustics and the structure of transient wave fields.** Michael G. Brown and Frederick D. Tappert (Department of Applied Marine Physics, University of Miami, Miami, FL 33149)

Transient acoustic wave fields in the vicinity of caustics in both two and three space dimensions are considered. In the vicinity of some possible caustics, the time history of the wave field due to an impulsive source includes a precursor. This noncausal behavior is reconciled with the physical principle of causality by requiring that such caustics be a part of certain higher dimensional caustics. With this constraint, all noncausal arrivals are preceded by a causal arrival and the total time history is causal. This constraint is used to derive some simple general properties of transient wave fields in the short wave limit, e.g., the number of arrivals must be an odd number.

1:50

**K2. Method for study of range-dependent time-differences-of-arrival between underwater sound eigenrays.** David C. Greene (Lexitech, 2289 Dosinia Court, Reston, VA 22091)

PASCAL language computer programs are described, which allow the calculation of eigenrays between an underwater source and receiver and the study of the range-dependence of time-differences-of-arrival (TDOAs) between eigenrays. The method uses a sound velocity profile from a disk file and user-supplied source and receiver depths. First, characteristics of rays which graze the surface and the bottom are calculated and presented to the user. Using this information, the user estimates eigenray angles. The program refines the user's estimate until the range error of the eigenray is smaller than a threshold value (typically 0.1 ft). Time of arrival for the eigenray is noted and the process is repeated for a set of range values. The process is repeated for other eigenrays. The range histories of arrival times of the eigenrays are used to calculate the range-dependent TDOAs. Using an 8-bit, 4-MHz microcomputer, less than 10 s are required to refine an estimate to an eigenray determination. Calculation of TDOA-versus-range data for ten pairs of five eigenrays takes about 30 min.

2:05

**K3. A simple wave-based model for unsaturated fluctuations.** Thomas B. Gabrielson (Naval Air Development Center, Warminster, PA 18974) and Suzanne T. McDaniel (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

Prediction of signal fluctuations, induced by sound-speed perturbations in a realistic, inhomogeneous environment in which diffraction is significant, requires the use of wave techniques. Ray-based models fail in regions of strong focusing and in regions where diffractive leakage is important. Consequently, a model has been developed based on the mode groups associated with the saddle points of the field integral for the unperturbed field and the eigenvalue shifts introduced by the perturbations. The results are directly applicable to acoustic propagation in either the atmosphere or the ocean.

2:20

**K4. Helmholtz integral formulation of the sonar equations.** Harry A. Schenck (Physics Department, U.S. Naval Academy, Annapolis, MD 21402-5026)

The sonar equations provide a convenient way to calculate the detection range of actual and proposed sonar systems. They are simple, but approximate, relationships between parameters which describe the effects of source, target, and acoustic medium independently. The purpose of this paper is to derive an exact form of the active sonar equation by using the Helmholtz integral formulation to solve the boundary-value problem with source and target both present in the medium. The resulting equation involves combinations of linear integral operators; however, it is suitable for solution by numerical techniques already developed for radiation from objects of arbitrary shape. Furthermore, it is shown that these integral operators reduce to multiplicative factors which represent general definitions of the source level, transmission loss, and target strength when the source-to-target distance is large. Thus this work establishes a basis for the sonar equations as the limiting form of an exact boundary-value problem and presents formulas for calculating the sonar parameters. [Work performed under a Naval Scientist Training and Exchange Program (NSTEP) assignment from the Naval Ocean Systems Center (NOSC).]

2:35

**K5. Calculation of underwater acoustic fields by Gaussian beam tracing.** Michael B. Porter (Naval Research Laboratory, Code 5120, Washington, DC 20375) and Homer P. Buckner (Naval Ocean Systems Center, Code 541, San Diego, CA 92152)

The method of Gaussian beam tracing has recently received a lot of attention in the seismological community. In comparison to standard ray tracing, the method has the advantage of being free of certain ray tracing artifacts such as perfect shadows and infinitely high energy at caustics. The technique is especially attractive for high-frequency problems or problems with complex bathymetry where normal mode, FFP, or parabolic models are not practical alternatives. The Gaussian beam method associates with each ray a beam with a Gaussian intensity profile normal to the ray. The beam curvature and width are governed by an additional pair of differential equations that are integrated along with the usual ray equations to compute the beam field along the ray. Analytic solutions to the beam equations are available for certain canonical problems, including the cases where the sound speed or square of the index of refraction is a linear function of the space variables. Thus ray tracing codes that employ patches of such domains are easily adapted to the beam tracing method. We present an overview of the method and results for certain ocean acoustic scenarios, which illustrate the performance of the method.

2:50

**K6. Bottom penetration from a range- and depth-dependent oceanic sound duct.** J. Miller, A. Nagl, and H. Überall (Department of Physics, Catholic University, Washington, DC 20064)

In a fundamental paper [F. B. Jensen and A. Kuperman, *J. Acoust. Soc. Am.* **67**, 1564 (1980)], the penetration of sound propagating upslope in a wedge-shaped ocean, into the ocean bottom, has been calculated using the parabolic-equation method; subsequently, this problem was treated by Evans [J. Acoust. Soc. Am. **74**, 188 (1983)] using the slab method, as well as by others. The mentioned work assumed an isovelocity water duct and isovelocity bottom. Previously [A. Nagl, H. Überall, A. J. Haug, and G. L. Zarur, *J. Acoust. Soc. Am.* **63**, 739 (1978)], we have developed an adiabatic mode theory approach to sound propagation in a range- and depth-dependent environment; this has now been generalized to include mode coupling. When applied to the Jensen-Kuperman wedge problem, we reproduce their results (downward directed bottom penetration). This is generalized by introducing sound-speed gradients, which leads to very important results: A positive (realistic) bottom gradient causes the bottom-penetrating sound to travel along the ocean floor, rather than downwards into the bottom. This effect has far-reaching consequences. [Work supported by the Office of Naval Research, and by the Singer Company—Link Simulation Systems Division.]

3:05

**K7. Some new results for ray transmissions in a wedge-shaped ocean.** T. H. Rousseau (Siena College, Loudonville, NY 12211), M. J. Jacobson, and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

The influence of a sloping bottom on cw ray transmissions in an isospeed channel is investigated. Bottom angles up to about  $3^\circ$  are considered, and regular perturbations off horizontal bottom results are employed. Accurate analytic approximations to ray-geometric quantities are derived. Then, sloping bottom influence on per-ray travel time and transmission loss is examined. Significant variations are shown to occur, such as travel time changes of more than 200 ms over ranges of about 6 km. Per-ray transmission loss is found to be strongly influenced by bottom slope. Variations of more than 20 dB are illustrated. Sloping bottom effects on the total acoustic field are examined also. A comparison of the influences on travel time of one model of a shallow-ocean front and the sloping bottom is made. The sloping bottom effect can induce travel time changes more than 300% larger than the frontal effect. [Work supported by ONR.]

3:20

**K8. Modified integral transform for weakly range-dependent shallow ocean Green's functions.** J. M. Arnold (Department of Electronics and Electrical Engineering, University of Glasgow, Glasgow G12 8QQ, Scotland) and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic Institute of New York, Farmingdale, NY 11735)

Acoustic Green's functions for a range-independent ocean and bottom environment can be expressed as integral transforms with respect to the range coordinate. This rigorous formulation contains all of the relevant source-excited wave phenomena (rays, normal modes with inclusion of continuous spectra, etc.) and is amenable, furthermore, to direct numerical treatment via the fast field program (FFP). Because of these features, it would be desirable to extend this formulation to range-dependent configurations. While the ability to do so in general is open to question, progress can be made for weak dependence on range. Refining our earlier plane wave spectral (intrinsic mode) study of the wedge-shaped ocean Green's function [J. M. Arnold and L. B. Felsen, *J. Acoust. Soc. Am.* **73**, 1105–1119 (1983)], we have been able to express the result as a range-dependent spectral transform that inverts an adiabatically approximated functional dependence on depth. This modified integral transform representation, which accounts for source effects, radiation into the bottom, uniform adiabatic mode transition through cutoff, etc., systematizes and generalizes the more heuristically derived form of Kamel and Felsen [J. Acoust. Soc. Am. **73**, 1120–1130 (1983)]. We present the new derivation and properties of this representation as well as its numerical implementation and its potential for extension to three-dimensional weakly range-dependent environments. [Work supported by ONR.]

3:35

**K9. Accurate PE calculations for range-dependent environments based on  $c_0$  updates.** Finn B. Jensen and Giovanna Martinelli (SACLANT ASW Research Centre, 19026 La Spezia, Italy)

The standard parabolic approximation to the acoustic wave equation is known to have intrinsic phase errors, which will degrade the accuracy of any PE solution for long-range propagation in the ocean. Pierce recently suggested a remedy to minimize these phase errors by simply choosing an appropriate mean phase speed ( $c_0$ ), eventually updated with range, representing a weighted average of all phase speeds involved in a particular propagation problem. We have now extended Pierce's formalism to include the wide-angle parabolic equation of Thomson and Chapman [J. Acoust. Soc. Am. **74**, 1848 (1983)], which inherently has smaller phase errors than the standard parabolic equation. The importance of using  $c_0$  updates in PE calculations for range-dependent environments is demonstrated through numerical results for a wedge-shaped ocean. Comparison with alternative solution techniques (coupled modes, intrinsic modes) shows that accurate PE solutions for the wedge problem can be obtained only for single-mode situations, even when using  $c_0$  updates with range.

3:50

**K10. A discussion on the angle of propagation in relation to the parabolic equation approximation.** Donald F. St. Mary (University of Massachusetts, Amherst, MA 01002) and Ding Lee (Naval Underwater Systems Center, New London, CT 06320)

In approximating the reduced wave equation by parabolic-like partial differential equations, it is interesting to observe that in the direction of propagation, the size of the angle is related to the order of the parabolic-like equation. The order of the Padé rational function approximation used in the derivation of the parabolic equation (PE) determines the order of the parabolic equation. The "standard PE" and the "wide-angle PE" can be thought of as having been derived through approximations by a Padé (1,0) and a Padé (1,1), and can be considered to have orders 2 and 3, respectively. In this paper, we discuss parabolic-like approximations using higher-order Padé rational functions which result in higher-order pseudopartial differential equations. Their behavior in relation to the angle of propagation is examined.

4:05

**K11. Direct generation of starting fields for parabolic equation models by perturbation theory.** E. Richard Robinson and David H. Wood (Naval Underwater Systems Center/New London Laboratory, New London, CT 06320)

It is well known that the standard approaches to the computation of parabolic equation model starting fields (such as Gaussian and normal modes) have definite drawbacks, with regard to running time or accuracy. In this paper, we present a method for computing the starting field which directly generates the combination of modes needed for the parabolic equation model's starting field. To demonstrate the speed and accuracy of the perturbation approach, we compute the starting field for a variety of system and environmental parameters. These results are compared to the more customary Gaussian and normal mode models.

4:20

**K12. Effects of random bottom structure and topography on transmissions in a shallow ocean.** C. E. Ashley, W. L. Siegmann, and M. J. Jacobson (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

In a recent paper [C. E. Ashley, M. J. Jacobson, and W. L. Siegmann, *J. Acoust. Soc. Am.* **76**, 1445–1455 (1984)], the authors studied the influences of stochastic horizontal bottom structure on underwater sound transmissions. Both bottom density and sound speed were taken to be random. Ray theory was used in an isospeed channel with horizontal

boundaries. In this study, we add to random structural effects those of a lossy bottom interface consisting of large-scale random facets with curvature, on which may be superimposed small-scale roughness. Each facet is assumed to possess small random slope, depth deviation, and curvature. Initially, we take acoustic rays to be specularly reflected from a large-scale facet bottom, and derive formulas for the mean and variance of incoherent intensity at a point receiver for a transmitted cw signal. The results are sufficiently general to permit their use with different bottom-acoustic models. Relative effects of structure and topography are compared. Subsequently, small-scale roughness is added to the facets, and the consequences of scattering are considered. [Work supported by ONR.]

4:35

**K13. The stabilization of stepwise coupled modes.** Richard B. Evans (Naval Ocean Research and Development Activity, NSTL, MS 39529)

A decoupling algorithm is derived to stabilize the calculation of the stepwise coupled modes. The stepwise coupled mode method reduces the elliptic wave equation in a discretely range-dependent environment to the solution of a matrix two-point boundary value problem. The solution of such boundary value problems can be unstable in the presence of modes whose amplitudes both grow and decay rapidly with range. This stability problem affects the accuracy of the stepwise coupled mode solution as range and frequency increase. The decoupling algorithm described here solves this stability problem. The algorithm makes it possible to decouple the growing solution from the decaying solution. Then each solution can

be computed accurately by integrating in a direction in which the solution decays. The stabilized stepwise coupled mode method is more accurate at longer ranges and higher frequencies than the original method. [Work supported by NORDA.]

4:50

**K14. Acoustical effects of ocean current shear in the parabolic approximation.** J. S. Robertson (U. S. Military Academy, West Point, NY 10996-1786), M. J. Jacobson, and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

In a previous study [J. S. Robertson, W. L. Siegmann, and M. J. Jacobson, *J. Acoust. Soc. Am.* 77, 1768-1780 (1985)], the authors developed a family of parabolic equations which include effects due to the presence of a steady, depth-dependent current. Three of these equations contained a new term which explicitly depended on current gradient. In this work, we study the effects of this new term. By appropriately transforming and simplifying, the equations are transformed into parabolic equations which can be integrated numerically with existing implementations. Presence of ocean current fine structure is one mechanism which can require new terms. Propagation is examined in a shallow isospeed channel with a lossy bottom and a variety of shear flows, some of which model actual ocean flows. Current fine structure can induce variations in intensity which are substantial and which depend on shear structure, source and receiver locations, and frequency. Finally, intensity differences are examined in reciprocal sound transmissions. [Work supported by ONR.]

TUESDAY AFTERNOON, 5 NOVEMBER 1985

SUITE 4, 1:30 TO 4:45 P.M.

## Session L. Shock and Vibration I: Acoustic Intensity: Experiments and Computations

Richard V. Waterhouse, Chairman

*David W. Taylor Naval Ship Research & Development Center, Bethesda, Maryland 20084*

Chairman's Introduction—1:30

### Invited Papers

1:35

**L1. Recent developments in acoustic intensity.** Gunnar Rasmussen (Brüel & Kjær, 18 Nærum Hovedgade, 2850 Nærum, Denmark)

The development of intensity measurement techniques has increased the demand for knowledge about the transducers used and for improved calibration techniques. The phase response of transducers is very important. Knowledge about the behavior of transducers at higher frequencies is especially significant for the determination of the effective acoustical spacing between transducers. The use of two, four, or six transducer probes is required in order to obtain the total energy density at a point. The estimation of the active intensity and the reactive intensity of a wave field depends on proper use of instrumentation and calibration technique and proper use of the transducers in a wave field. The configuration in which the transducers is mounted plays an important role in obtaining good data. Development in instrumentation enables the use of gating techniques, which improves the signal enhancement. By combining information in frequency and time domain a better understanding of source behavior is achieved.

2:00

**L2. Use of the complex intensity vector for sound radiation and sound field studies.** Jiri Tichy and Adin Mann (Graduate Program in Acoustics, The Pennsylvania State University, State College, PA 16804)

The traditional intensity technique to determine the sound power radiated by sources or propagating through a sound field uses the real part of the complex intensity only. More information on the sound field and sound radiation can be obtained if both real and imaginary components of the intensity vector are considered. This paper presents the results of studies of the fundamental relationships between sound pressure, particle velocity, complex intensity, and both potential and kinetic energy. As shown before, the real part of the

intensity vector is rotational ( $\text{curl } I_r \neq 0, \text{div } I_r = 0$ ), while the imaginary (reactive) component of the complex intensity vector is irrotational ( $\text{curl } I_i = 0, \text{div } I_i \neq 0$ ). The conditions of the vortex formation were studied as a function of the source distribution. The relation between the extremes of the intensity curl and the energy in the sound field have been formulated. The applications of the sound intensity field structure on the sound radiation from complex radiators is shown using specific examples. New graphical methods of presenting the intensity field are shown. The paper also presents experimental results obtained by an automatic computer controlled system for intensity measurements.

2:25

**L3. Measurement of the acoustic intensity in the nearfield of a fluid-loaded, point-driven cylinder using nonplanar nearfield acoustical holography.** Earl G. Williams (Naval Research Laboratory, Code 5133, Washington, DC 20375-5000)

Nearfield acoustical holography (NAH) is confined to the reconstruction of acoustic fields in parallel planes with a hologram consisting of a measurement of the acoustic pressure over a plane boundary. Although ideal for planar sources, such as vibrating plates, NAH encounters problems in dealing with nonplanar sources. Therefore, we have modified the technique to apply specifically to a cylindrical geometry with a hologram comprising a cylindrical contour. The cylindrical hologram consists of the measurement of the amplitude and phase of the pressure field at 4096 points on this cylindrical contour. This contour is located close, and concentric with, a radiating cylindrical source (in this case a finite, point-driven cylindrical shell submerged in our underwater tank facility). We show how this hologram is processed in a rigorous manner to map the pressure, vector velocity, and vector intensity (real and reactive components) from the surface of the source into the farfield. The vector intensity maps are shown to be helpful, in many cases, in identifying the location of the point driver attached inside the shell. Both the active and reactive intensity fields will be shown. Since the vector velocity is also obtained from the cylindrical hologram, we can identify the radial mode shape of vibration from the radial velocity component at the shell surface, and along with the reconstructed surface pressure (which provides the fluid loading) we can also map the surface intensity. This latter quantity is used to compute the total power radiated by this mode shape. This new technique provides a comprehensive method to study radiation from cylinders.

2:50

**L4. Energy streamlines for sound sources.** R. V. Waterhouse and D. Feit (David Taylor Naval Ship Research & Development Center, Bethesda, MD 20084-5000)

These streamlines are constructed so that at each point on one, the sound intensity vector is tangential to the streamline. They thus show the direction of flow of the sound energy in an acoustic field. Energy streamlines are formally analogous to the velocity streamlines familiar in fluid mechanics, and for axially symmetric sources represent contours of the Stokes stream function. The latter function can be computed from the intensity function, as the two are analytically related in a skew differential manner. Examples of the streamlines are given for the circular piston source in an infinite rigid baffle and the water-loaded plate driven by a line or a point source. The streamlines are chosen so that equal energy flows between each adjacent pair of streamlines. The sound pressure field is irrotational, but the sound intensity field is rotational, and interesting patterns of circulating energy, including vortices, occur for some source configurations.

### Contributed Papers

3:15

**L5. Some problems associated with making sound intensity measurements in the presence of reverberant fields and/or background sources.** A. F. Seybert and J. A. Holt (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506-0046)

This paper is concerned with potential problems that one may encounter when attempting to use the two-microphone method to determine the sound intensity and sound power of a source when background and reverberant fields are present. Under such conditions, the primary variables controlling the accuracy of the sound intensity estimate are shown to be the ratio of the unknown intensity to the mean-square pressure of the background/reverberant field and the accuracy of the estimate of the phase angle between the two microphones. Proceeding from a few simple concepts we illustrate the abovementioned measurement problem with results obtained from a controlled experiment. [Work supported by International Business Machines Corporation.]

3:30

**L6. A nonlinear coherence function and its application to machine diagnostics.** Tom Coffin and Jen Yi Jong (Wyle Laboratories, P. O. Box 1008, 7800 Governors Drive West, Huntsville, AL 35807-5101)

The harmonic content in dynamic measurements from rotating machinery contains much subtle information concerning equipment operational condition and component degradation. For this reason, the power density spectrum (PSD) has long been employed to assess the relative magnitude of fault-related spectral contributions. Measurements on high-performance rocket engine turbomachinery suffer from severe noise contamination from numerous extraneous sources, which impedes rotating element diagnostic evaluation. It is thus difficult to determine whether an apparent high-level harmonic contribution is indeed related to the fundamental rotational frequency,  $f_r$ , or possibly due to an independent source. The ordinary PSD, being an absolute value, is of no assistance to this problem. In an effort to relate synchronous speed characteristics with an

arbitrary harmonic component, an unique coherence spectrum was devised which we call the *hyper-coherence* function. The hyper-coherence function,  $H_n(f_i)$ , defines the nonlinear correlation between waves at the fundamental frequency and harmonics at  $nf_i$ ,  $n = 1, 2, \dots$ . The computation is straightforward by FFT methods, and results in a line spectrum of correlation coefficients as a function of harmonic number. This paper presents the hyper-coherence as a multiple-ordered correlation function and illustrates its utility through application to several idealized signals, and the assessment of an apparent high-amplitude harmonic signature in rocket engine vibration measurements.

3:45

**L7. Acoustic intensity of scattered field from a rib-reinforced plate.** Cetin Seren and Sabih Hayek (Department of Engineering Science and Mechanics and the Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

An infinite elastic plate which is reinforced by one rib is insonified by a plane incident wave. The diffracted acoustic pressure generated by the rib welded to the plate is computed by an asymptotic series and by using a new integral representation which converges very rapidly. The acoustic intensity in the nearfield and farfield of the plate are computed and exhibited as an intensity map. The intensity map shows the propagation of the diffracted acoustic energy from the rib and the local energy interchange with the elastic plate in its vicinity. [Work supported by NAVSEA.]

4:00

**L8. Surface acoustic intensity distributions on vibrating plates driven by distributed forces.** R. F. Keltie and H. Peng (Center for Sound and Vibration, Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910)

The two-dimensional acoustic intensity distributions on the surface of vibrating plates driven by multiple point forces and by finite-sized force distributions have been examined analytically and experimentally. Calculations and measurements were made using a 5- $\times$ -5-ft. aluminum plate of 0.25-in. thickness, simply supported around the periphery. Results are shown for the specific case of two harmonic point forces, acting at numerous different locations on the plate's surface. One parameter of interest concerns the relative phasing between the two forces, and computational results are presented for the cases of in-phase, 90° out-of-phase, and 180° out-of-phase forces. In addition, acoustic intensity patterns associated with disk force and line force excitations are presented. The results are discussed in terms of frequency and in terms of the relative importance of

different portions of the plate, including the regions near the forces, and the regions corresponding to edge mode and corner mode radiation. [Work supported by NSF.]

4:15

**L9. Power flow and traveling waves in structures.** Albert J. Tucker and Joseph A. Clark (David W. Taylor Naval Ship R&D Center, Bethesda, MD, 20816 and Catholic University, Washington, DC 20064)

Recent experiments [J. A. Clark and A. J. Tucker, "Optical Measurements of Structural Intensity Distributions," 2nd Int. Congress on Acoustic Intensity (Sept. 1985)] have revealed that the traveling waves associated with power flow in structures can be separated from the standing waves (mode shapes) associated with vibrational energy stored in structures. The separation is achieved by an optical measurement method which employs phase-resolved holographic interferometry to determine the instantaneous deformed shape of a structure when it is in quadrature phase relationship to the force driving the structure. In this talk, a theoretical analysis of the correspondence between power flow and the coincident and quadrature shapes of vibrating structures will be presented and the theoretical results will be compared with features of traveling and standing waves visualized in an actual vibrating structure. [Research supported by DTNSRDC and ONR.]

4:30

**L10. Jet noise source location via acoustic intensity.** Werner G. Richarz (Department of Mechanical and Aeronautical Engineering, Carleton University, Ottawa, Ontario, Canada K1S 5B6)

It is well known that sources of jet noise are described by a variety of models; thus there is no unique source distribution. If no assumptions are made about the nature of the sources, then one can infer virtual source distributions from certain farfield measurements. Here, however, the source locations are deduced directly from acoustic intensity measurements near an axisymmetric jet flow. The magnitude and direction of the acoustic intensity vector are found to be functions of position and source frequency. Acoustic intensity is measured along three directions in a plane of the jet axis; data are analyzed in 1/3 octave bands. The measurements are performed sufficiently far from the jet flow to avoid air flow over the probe. Clearly, the resolution of the source location scheme suffers with increasing probe-source separation: Thus a compromise is struck. The measured data permit the reconstruction of certain properties of the acoustic farfield and appear to demonstrate closure. [Work funded by Natural Sciences and Engineering Research Council of Canada.]

TUESDAY AFTERNOON, 5 NOVEMBER 1985

DAVIDSON ROOM B, 2:00 TO 4:45 P.M.

## Session M. Physical Acoustics II: Sound Generation and Radiation

Peter H. Rogers, Chairman

*Department of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332*

### Contributed Papers

2:00

**M1. Underwater sound generation by a high-intensity laser pulse.** Nicholas P. Chotiros (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

In the generic optoacoustic sound source, a laser beam is directed at the water surface to produce a controlled local reaction which, in turn, generates sound waves. The energy is delivered without physical contact. At low optical intensities, the reaction is a linear thermal expansion of the

medium, but at high intensities, the reaction is nonlinear and more complicated. A simple blast model is used to predict the acoustic output of the high intensity reaction. The blast model is based on that of A. N. Pirri who successfully used it to model the momentum transfer between a high power laser blast on the surface of a solid. The Pirri model is general enough to be adapted for laser-induced blasts over water. The model predictions are compared with experimental results. The differences are discussed. [Work supported by ONR.]

**M2. Experiments with sound induced by a laser pulse moving at Mach 1 over a water surface.** Yves H. Berthelot and Ilene J. Busch-Vishniac (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

Sound generated in water by a laser via thermal expansion, is usually of very low amplitude. However, experimental results show that, when the laser beam is moving on the surface of the water at velocities close to the speed of sound in water, a much stronger acoustic pulse is generated and propagates underwater. The linear theory describing the radiation of sound by thermal expansion underestimates significantly the acoustic levels observed experimentally in such a Mach wave. Experimental data also include pressure waveforms, directivity patterns, sound level dependence on source velocity, and distance of observation. The lasing wavelength dictating the penetration depth of light in water, could be either  $1.06\ \mu\text{m}$  (Neodymium:Glass) or  $0.6943\ \mu\text{m}$  (ruby).

2:30

**M3. Asymptotic analysis of diffraction effects in the radiation of underwater sound by interesting configurations of laser-generated heat deposition.** Hsiao-an Hsieh and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A previously proposed ideal configuration [J. Acoust. Soc. Am. Suppl. 1 77, S104 (1985)] for optimal laser-generated heating of water to achieve strong farfield acoustic signals with desirable properties suggests a number of pertinent boundary value problems in thermoacoustics. These progress in complexity and are analyzable with varying degrees of simplicity and thoroughness using mathematical physics techniques such as Fourier transforms, contour deformation, and the saddle point method. A simple class of such models assumes that the source term (associated with heat addition) of the wave equation is independent of the horizontal coordinate  $y$ , is a sinusoidal progressing wave in the horizontal coordinate  $x$ , and dies out exponentially with depth coordinate  $z$ . The source extent in the  $x$  direction is either taken as unlimited in the simplest idealization or non-zero only over a fixed length  $L$ . The pressure disturbance is described as a Fourier integral over wavenumber  $k_x$  and angular frequency  $\omega$ . The ordinary differential equation for the integrand is readily solved, and the resulting integrals are evaluated for positions and times of interest using complex variable techniques. [Work supported by ONR, Code 425-UA.]

2:45

**M4. Green's function analysis of free shear layer instabilities in jets of arbitrary cross section.** Shozo Koshigoe (Engineering Science Division, Code 3892, Naval Weapons Center, China Lake, CA 93555) Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

Sound generation via the Kelvin-Helmholtz instability mechanism in jet shear layers is closely related to instability modes with the largest spatial growth rates [A. Michalke, J. Fluid Mech. 23, 521-544 (1965)]. Recently, there has been considerable experimental interest in noncircular jets [e.g., E. Gutmark *et al.*, AIAA Shear Flow Control Conf., 12-14 March, Colorado Springs, CO, 1985; and R. A. Peterson *et al.*, Bull. Am. Phys. Soc. 29, 1562 (1984)]. In order to theoretically describe these cases, we have used Green's function techniques to develop practical solution methods for the Orr-Sommerfeld linearized stability equations with arbitrary jet mean flow profiles. Some preliminary results are presented for critical Reynolds numbers, instability mode eigenfrequencies, and shape functions. [Work supported by NWC research grant.]

3:00

**M5. A sound source near a velocity discontinuity in the presence of a rigid boundary.** Mauro Pierucci and Nagy Nosseir (Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

The problem of the interaction between an acoustic source and a velocity discontinuity is a problem of interest to researchers in different areas. Gottlieb [J. Acoust. Soc. Am. 32, 1117-1122 (1960)] has solved for the farfield acoustic radiation transmitted through a discontinuity. In this paper, the velocity discontinuity is assumed to impinge on a rigid surface so that the acoustic interaction is now between the source, the velocity discontinuity, and the rigid wall. Displacement of the interface has been evaluated numerically by the use of a fast Fourier transform subroutine on a CYBER 750 computer. Variation of the displacement as a function of the acoustic source location and of the mean flow velocities of the two media is presented. The results indicate that the interface displacement is very sensitive to the location of the sound source. For given flow conditions two optimum sound sources exist which will maximize the interaction between the sound source and the fluid discontinuity. For all cases run, the spatial variation of the displacement of the fluid interface is seen to be composed of the superposition of two wavelengths; the long wavelength is related to the acoustic wavelength while the shorter wavelength is a function of the local flow field conditions.

3:15

**M6. Transient acoustic radiation and scattering from one-dimensional fluid-loaded vibrators via impulse response methods.** Duraisingh D. Ebenezer and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881-0814)

A general time domain approach is presented to evaluate the transient velocity and pressure field of fluid-loaded one-dimensional vibrators, which are excited by broadband excitations. The approach is based on the use of time-dependent modal expansion for the velocity of the vibrator. As a result of the fluid coupling, the time-dependent coefficients of the modes are described by a set of coupled convolution integral equations which are solved for a specific excitation by marching forward in time. A time-dependent Green's function method is used to obtain a convolution integral representation for the pressure field. Numerical results will be presented for different types of one-dimensional vibrators which include both plate and beam configurations. Differences in the nature of the fluid coupling for the two configurations of vibration will be highlighted in the time domain. Numerical results will illustrate the importance of the modal coupling for both configurations. [Work supported by ONR.]

3:30

**M7. Alternative variational principles for sound radiation by vibrating structures.** Allan D. Pierce and X.-F. Wu (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Desirable features of variational principles are discussed and arguments are presented against indiscriminate uses of Galerkin's method that proceed from non-self-adjoint formulations. An ideal principle appears as a stationary functional for a quantity of direct physical interest, such that a mediocre estimate of trial functions results in a good estimate of that quantity. The authors' previously derived variational formulation, based on the Kirchhoff-Helmholtz surface integral corollary, is shown to yield a variational expression for a weighted average radiation impedance. Variational expressions for other quantities of physical interest are derived using the general technique of Gerjuoy *et al.* [Rev. Mod. Phys. 55, 725-774 (1983)]. Such includes the product of farfield pressure and radial distance in any specified direction, because this is expressible as a constant plus a weighted surface integral over pressure. The derived variational functional involves not only a trial function for the surface pressure for the originally posed problem, but also one for a secondary problem which usually has a clear physical interpretation. Analogies with scattering theories ensuing from the 1940s work of Schwinger and Levine are discussed and it is argued that advances in computer technologies make it desirable to reexamine such theories' acoustical applications. [Work supported by ONR.]

3:45

**M8. A simplified boundary element formulation for acoustic radiation and scattering for axisymmetric bodies and boundary conditions.**

A special formulation involving axisymmetric bodies and boundary conditions is presented using the boundary element method. For this special case, the surface integrals are reduced to line integrals and an integral over the angle of revolution. The integration over the angle is performed partly analytically in terms of elliptic integrals and partly numerically using simple Gaussian quadrature formula. This model allows simple discretization scheme for a desired degree of accuracy and less computing time as compared to the general three-dimensional boundary integral equation method. Examples are shown for problems involving spheres and cylinders.

4:00

**M9. Finite element implementation of a variational formulation for acoustic radiation and scattering exemplified by the circular disk.** J. S. DiMarco and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A previously derived variational principle [A. D. Pierce and X. -F. Wu, *J. Acoust. Soc. Am. Suppl.* 1 74, S107 (1983)] for the acoustic perturbation on the surface of a body, either vibrating or acting as a rigid-body scatterer, is a possible starting point for finite element formulations. The present paper studies the particular example of a circular disk that is presumed rigid and axisymmetrically excited; surface pressure is assumed to vary linearly between mesh points. An integrated Green's function that represents the signal at a field annulus due to a source annulus is pretabulated and interpolated in the computation scheme. The formulation yields a system of simultaneous equations for the pressures at the mesh points. Computation of coefficient matrix elements requires a twofold integration, with the axisymmetric annular Green's function in the integrand, and with the integration limits corresponding to the inner and outer edges of two annular regions. Numerical techniques for the integrations are discussed and the effects of mesh spacing and of numerical interpolation of the annular Green's function are explored. Results are compared with computations involving spheroidal wavefunctions and with computations using a Rayleigh-Ritz scheme based on the same variational formulation. [Work supported by ONR.]

4:15

**M10. Influence of baffle size and shape on farfield radiation patterns.** Shokichi Tanaka (Laboratory, Japan Radio Co., Ltd., 1-1, Shimorenjaku 5-chome, Mitaka-shi, 181, Japan)

In underwater acoustics, transducers are often placed in some kind of a baffle. Then the farfield radiation patterns of the transducers vary with the property of the baffle and its size and shape. This paper describes the basic characteristics of the influence of a baffle due to its size and shape. Two shapes of baffles are treated to investigate the influence: (1) a three-sided semi-infinite baffle with a piston flush mounted in its middle side and (2) a semi-infinite circular cylinder baffle with a piston in its end side, both having either rigid or soft property. The geometrical theory of diffraction is used to calculate the farfield radiation patterns of the piston in the case of the middle side and the method of equivalent edge acoustic sources in that of the end side. Numerical results obtained with the methods in various baffle sizes and piston locations are presented. Measured patterns, which agree well with calculated ones in soft condition, are also presented.

4:30

**M11. Gaussian-Laguerre formulation of Lommel diffraction problem.** Bill D. Cook (Cullen College of Engineering, University of Houston, University Park, Houston, TX 77004)

Solutions of Fresnel diffraction problems based on Gaussian-Laguerre functions are easily interpretable and can be calculated quickly because of recursive relations between the functions. I have applied them to solving the diffraction problem of a circular, receiving piston transducer in the field of a similar sending transducer. The pistons are coaxial but not restricted to being the same size. For equal size transducers, I find excellent agreement with earlier results [P. H. Rogers and A. L. Van Buren, *J. Acoust. Soc. Am.* 55, 724-728 (1974)]. When the receiving transducer becomes very small, the answer on axis agrees with the analytic point value. When the receiver has a radius twice as large as the sender, the Lommel diffraction correction is nearly a constant over a large range. [Work performed under the auspices of the U. S. Department of Energy and Lawrence Livermore National Laboratory, Contract No. W-7045-ENG-48.]



**Session N. Underwater Acoustics III: Signal Processing**

Arthur B. Baggeroer, Chairman

*Department of Ocean Engineering, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139***Chairman's Introduction—8:00****Invited Papers****8:05****N1. Spectral analysis via lag reshaping of the correlation estimate.** Albert H. Nuttall and G. Clifford Carter (Naval Underwater Systems Center, New London, CT 06320)

The possibility of achieving maximally stable, low-sidelobe spectral estimates, without the need for overlapping or temporal weighting, is investigated both theoretically and via simulation. In particular, the (frequency domain) power spectral estimates of each of a sequence of abutting rectangularly gated time data segments are averaged, and then Fourier transformed into the lag (or correlation) domain. This correlation estimate is then reshaped, by dividing out the undesirable triangular autocorrelation of the rectangular temporal weighting, and by multiplying by a desirable lag-weighting function with low sidelobes. Another Fourier transform yields the final spectral estimate of interest. This technique includes, as special cases, the Blackman-Tukey technique and the weighted overlapped segment-averaging FFT technique. The general method has been analyzed in terms of the mean and variance of the spectral estimate, thereby revealing the fundamental dependence of its performance on the temporal weighting, lag weighting, amount of overlap, number of pieces, available data record length, and desired frequency resolution. The mean spectral estimate is equal to the convolution of the true spectrum with an effective window. In the case of lag reshaping, the effective window corresponds directly with the desirable lag-weighting function above, with its low sidelobes. More generally, the effective window is equal to the convolution of the lag window with the magnitude-squared temporal window. Analytic results for the variance of the spectral estimate with rectangular temporal weighting indicate that if the length of the temporal weighting is selected to be somewhat larger than the length of the lag weighting, the variance is at a near minimum. Furthermore, in this situation, the possibly deleterious sidelobes of the temporal weighting can be exactly compensated by proper choice of lag weighting, resulting in low sidelobes and good decay of the overall effective spectral window. Simulation results that confirm all these effects predicted theoretically are presented. The possibility of detecting a weak tonal via lag reshaping is demonstrated, both for a nearby frequency as well as a distant tonal.

**8:30****N2. Experimental performance of single and multiple-channel high-resolution spectral estimators.** S. L. Marple (Schlumberger Well Services, 5000 Gulf Freeway, Houston, TX 77252)

This presentation will examine the performance characteristics of a wide variety of spectral estimators, including the techniques of MEM, Prony, AR (auto regressive), minimum variance or MLM, and eigenanalysis (singular value decomposition and multiple signal classification), to which high-resolution behavior has been attributed. A brief tutorial will review the basis for each technique. The performance of each will be depicted through a series of test cases of different signal classes, e.g., broadband, tonals, sunspots. The purpose of this is to demonstrate the strong and weak attributes of each method discussed.

**8:55****N3. Array processing and underwater acoustics.** H. Cox (Bolt Beranek and Newman Inc., 1300 N. 17 Street, Arlington, VA 22209)

Many array processing algorithms for underwater acoustics now have been developed. Particular emphasis has been on obtaining both high resolution with small apertures and number of sensors and adaption to environmental noise fields. With the availability of high-speed and small digital signal processors many of these algorithms can now be implemented, often in real time. This presentation will review these algorithms and their effectiveness in underwater acoustics.

**Contributed Papers****9:20****N4. Locating underwater sources of sound, using sensors on a 3-D line of unknown shape.** Homer P. Bucker (Naval Ocean Systems Center, Code 541B, San Diego, CA 92152)

If sensors are located on a line of known shape, underwater sound sources can be located by matched-field processing [H. Bucker, J. Acoust.

Soc. Am. **59**, 368-373 (1976)]. That is, the sources are associated with relative maxima of a correlation function  $c(r, z, \phi)$  which matches the received acoustic signals with calculated fields for source locations at different ranges ( $r$ ), depth ( $z$ ), and bearings ( $\phi$ ). If the line shape is not known, the methods of linear programming [H. Bucker, J. Acoust. Soc. Am. **63**, 1451-1454 (1970)] can be used to find an "effective shape" that corresponds to maximum sharpness ( $s$ ) of the "field of vision." Here,  $s$  is the

sum of  $c^2$  taken over all values of  $r$ ,  $z$ , and  $\phi$ . Examples will be presented showing sensitivity of the method to errors in the propagation model and to the effect of sources lying outside the matched-field points. [Work supported by NAVOCEANSYSCEN IR Program.]

9:35

**N5. Source localization using the PE method.** F. D. Tappert, L. Nghiem-Phu, and S. C. Daubin (Daubin Systems Corporation, 104 Crandon Boulevard, Suite 400, Key Biscayne, Miami, FL 33149)

The most important information for source localization is oceanographic knowledge supplied to a powerful machine that numerically computes acoustic Green's functions rapidly and accurately. By this means the distorting effects of the oceanic medium and its boundaries can be removed, thereby rendering the ocean transparent. Then targets can be detected, localized, and classified at low S/N ratios as though they were moving in free space, leading to the result there is "no place to hide." As an illustration of this approach, we discuss in detail the solution of the problem of passive narrow-band acoustic localization (an instance of the inverse source problem) following the pioneering work of Bucker [J. Acoust. Soc. Am. 59, 368-373 (1976)], in which beamforming was made obsolescent. Using the PESOGEN (Parabolic Equation *SO*lution *GEN*erator) computer system [J. Acoust. Soc. Am. Suppl. 1 75, S26 (1984)] and an algorithm based on the generalized principle of reciprocity, we have demonstrated theoretically that targets can be accurately and reliably localized at long ranges and low S/N ratios using sparse configurations of sensors.

9:50

**N6. Source localization in a waveguide using a matched-field technique.** C. Feuillade (ODSI Defense Systems Inc., 6110 Executive Boulevard, Rockville, MD 20852) and Wayne A. Kinney (Naval Ocean Research and Development Activity, Code 240, NSTL, MS 39529)

The ability to localize a source in a waveguide by correlating measured and predicted acoustic field values at a vertical array of receivers depends at least on (1) the number of dominant acoustic modes that have been adequately sampled and (2) the accuracy of the field predictions. Several attempts to localize sound sources, using experimental and model data for shallow and deep water environments, are discussed. The results of these attempts are presented as well as observations regarding the number of depth/range replica source points required to localize the source for complicated (many modes) versus simple (few modes) acoustic environments. [Work supported by NORDA.]

10:05

**N7. Coherence versus time delay for spatially separated receivers in deep water using a z-transform method.** C. Feuillade (ODSI Defense Systems Inc., 6110 Executive Boulevard, Rockville, MD 20852) and Wayne A. Kinney (Naval Ocean Research and Development Activity, Code 240, NSTL, MS 39529)

Broadband coherence values versus time delay for two horizontally separated receivers and for different source azimuth angles (from broadside to endfire) are presented for (a) a surface-bottom (SB) path, (b) a bottom-surface (BS) path, and (c) a combination of SB and bottom-reflected (B) paths. The water depth and range are both 5 km. The source and receivers are all at 200-m depth. The ocean bottom is assumed to be rigid and flat. Scattering from the ocean surface is modeled with the facet-ensemble method [H. Medwin, J. Acoust. Soc. Am. 69, 1060-1064 (1981), W. A. Kinney and C. S. Clay, J. Acoust. Soc. Am. (submitted)] applied to a long-crested wave model of the ocean surface using actual wave-height measurements with an rms value of 1 m. A z-transform algorithm [suggested by C. S. Clay (private communication)] has been used to perform broadband filtering and to compute coherence. Results demonstrate how signals propagating via the SB path possess greater coherence (with values near 0.8) than signals propagating via the BS path (values near zero). [Work supported by NAVELEX 612.]

10:20

**N8. Active and passive imaging of features on a rough seafloor.** C. S. Clay and M. R. Daneshvar (Geophysical and Polar Research Center, University of Wisconsin, Madison, WI 53706)

We are exploring array design and signal processing techniques for imaging features on the seafloor. In our numerical studies, rough features are modeled as diffracting wedges. Active imaging uses known sources. Passive imaging uses natural sources such as storms, earthquakes, etc. Natural sources are modeled as occurring at random times and random locations. We use delay and sum array processing to focus the array on diffraction sources. The results are displayed as two-dimensional vertical and horizontal pictures of diffraction sources. We also compare two methods of processing, squaring the array output and split array cross correlation. The diffractograms show that a wedge can be identified for active and passive sources. [Work supported by Weeks Bequest, Wisconsin Alumni Research Foundation, National Science Foundation, and NORDA.]

10:35

**N9. Reconstruction of a complex acoustic field from its real or imaginary part.** Michael S. Wengrovitz (MIT/WHOI Joint Program in Oceanography/Oceanographic Engineering, Woods Hole, MA 02543), George V. Frisk (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), and Alan V. Oppenheim (Research Laboratory of Electronics, Department of EECS, Massachusetts Institute of Technology, Cambridge, MA 02139)

In underwater acoustics, the sound pressure field in a horizontally stratified, range-independent medium due to a continuous-wave point source can be described by a Hankel transform of the depth-dependent Green's function. Although the total field consists of both outgoing and incoming components, a reasonable assumption is that the incoming components can be neglected. This assumption is the basis for a number of synthetic field generation techniques such as the Fast Field Program (FFP) [F. R. DiNapoli and R. L. Deavenport, J. Acoust. Soc. Am. 67, 92-105 (1980)]. It is shown that the condition that the field consists only of outgoing components implies a relationship between the real and imaginary parts of the field. An implication is that the real (imaginary) part of the pressure field can be reconstructed from the imaginary (real) part. An algorithm for performing this reconstruction is presented and examples of its application to synthetic and experimental acoustic fields is discussed.

10:50

**N10. Effects of convergence zone multipath on cross correlation.** Walter S. Hauck, III, E. Richard Robinson, and Azizul H. Quazi (Naval Underwater Systems Center, New London, CT 06320)

Recent work by Hauck [NUSC TD 7409] indicates autocorrelation processing may be adversely affected by the large number of acoustic rays from a source within a convergence zone. The object of the current study is to examine the effect of CZ multipath on cross correlation of frequency hopped pulses. The sensitivity of the correlation to source or receiver depth, and the location of the source within a convergence zone is examined using the generic sonar model [H. Weinberg, NUSC TD 5971C]. Results for both first and second CZs will be presented.

11:05

**N11. Autoregressive modeling of nonstationary multibeam sonar reverberation.** James C. Luby and Dean W. Lytle (Applied Physics Laboratory, University of Washington, 1013 NE 40, Seattle, WA 98105)

This work describes a technique for simulating nonstationary, multi-beam Gaussian sonar reverberation sequences. The primary objective is to simulate quadrature-demodulated complex digital vector sequences that have a prescribed time variant covariance function. We use a first-order integral scattering model to relate the geometrical, environmental, and sonar parameters to the assumed locally stationary multivariate reverberation power spectrum for each of a set of ranges spanning the total observation range of interest. At each range we compute the multivariate autoregressive canonical factorization of the power spectrum. The resultant autoregressive models are used to generate multivariate, correlated, sta-

tionary realizations for each range. Next we combine stationary realizations adjacent in range with an overlapped windowing scheme. This scheme allows the spectral characteristics of the reverberation to change with time. Finally, a time variable gain is applied to the entire realization to adjust the average intensity.

11:20

**N12. Triangulation ranging utilizing a conditional beta input error density.** J. J. Perruzzi, E. J. Hilliard, Jr., and D. Pereira (Naval Underwater Systems Center, Newport, RI 02841-5047)

Triangulation ranging is a method of estimating range to an acoustic source by utilizing the measured difference in arrival time (time delays) of spherical wave fronts at two receiving sensors. Errors in the measured time-delay values yield inaccurate range estimates. If these errors are unrealistically large, negative range values will result, and in practice are eliminated. A symmetrical beta density function is employed to model time-delay measurement errors. In order to insure that all range estimates are physically realistic, the end points of this beta density are clipped, which leads to a conditional distribution. The resulting distribution is subsequently used to obtain the triangulation range density along with its mean and standard deviation.

11:35

**N13. Multipath effects in a nonisotropic medium on a wave front curvature ranging system.** P. Bilazarian (Raytheon Co., Submarine Signal Division, P. O. Box 360, Portsmouth, RI 02871)

The influence of multiple ray arrivals on the localization performance of the wave front curvature ranging (WCR) system is investigated. The novelty of this work is that this influence is analyzed for a nonisotropic medium and for a system which performs time-delay measurements with limited resolution capabilities. The WCR system consists of three sensors located on a horizontal line. Source location parameters are determined from estimates (via cross correlation) of time delays associated with the aggregate of ray arrivals at each sensor. Conditions are defined where localization accuracy is affected by multipath arrivals. Formulas are developed which predict localization errors in terms of multipath time-delay differences between various sensor pairs. These formulas incorporate the system discrimination capabilities and contain multipath parameters, such as amplitude and arrival-angle differences. The effect of multipath on system performance is demonstrated for a variety of sound-speed profiles and multipath environments.

WEDNESDAY MORNING, 6 NOVEMBER 1985

SUITE 4, 8:15 TO 11:40 A.M.

WEDNESDAY AM

## Session O. Noise II: Annoyance Caused by Low-Frequency Noise Sources

Paul D. Schomer, Chairman  
U.S. Army CERL, P.O. Box 4005, Champaign, Illinois 61820

### Invited Papers

8:15

**O1. Low-frequency noise problems and psychological and physiological influences of low-frequency noise.** Shinji Yamada (Yamanashi University, Takeda 4 Kofu 400, Japan), Toshio Watanabe (Fukushima Technical College, Japan), and Toshifumi Kosaka (Tokyo Technical College, Japan)

In Japan infrasound or low-frequency noises are emitted from highway bridges, large compressors, boilers, and diesel engines etc. Windows or doors of wooden houses are vibrated by infrasound and emit rattling noises. Low-frequency noises are sometimes heard directly and annoy people. These complaints and low-frequency noises were investigated. Psychological and physiological influences on university students and low-frequency noise sufferers were investigated in a laboratory. University students were not influenced as much by low-frequency noise. The results clearly indicated that the heart and respiration rates of several low-frequency noise sufferers increased. Many low-frequency noise sufferers were influenced psychologically or physiologically.

8:40

**O2. Community response to low-frequency noise from large wind turbine generators.** Kevin P. Shepherd (The Bionetics Corporation, 20 Research Drive, Hampton, VA 23666)

The introduction of large wind turbine generators in the United States and Europe has, in some instances, resulted in adverse community reaction to noise and noise-induced building vibrations. In particular, this has been associated with horizontal axis wind turbines for which the rotor blade is mounted downwind of the supporting tower. Such an arrangement results in the generation of low-frequency impulsive noise due to the passage of the rotor blades through the tower wake. This paper will review the available literature on community reaction to low-frequency wind turbine noise. In particular, levels of noise and noise-induced building vibrations will be estimated by using available measured data and a recently developed low-frequency sound propagation model. Estimated levels will be compared with known perception thresholds of low-frequency noise and building vibrations. [Work supported by NASA Langley Research Center.]

9:05

**O3. Community response to low-frequency impulsive sources.** Paul D. Schomer (U.S. Army Construction Engineering Research Laboratory, P.O. Box 4005, Champaign, IL 61820)

Most army noise problems are the result of two types of sources: helicopters or large weapons (artillery, etc.). Through analysis of complaints and through attitudinal surveys, the U.S. Construction Engineering

Research Laboratory (USA-CERL) has studied community response to these sources. The results show that large-amplitude impulsive noise is better assessed using C-weighting rather than A-weighting, and that building vibration and rattles are the primary adverse factors. Based on these results, USA-CERL, with FAA support, studied the role noise-induced vibrations and rattles play in human response to helicopter noise. A "laboratory" test was performed in the field, using almost 200 subjects. The test used paired comparisons in two real dwelling units. The source as a UH-1H (Huey) helicopter and the control was 500-Hz octave band of white noise with an amplitude (envelope) temporal pattern shaped to approximate a helicopter fly by. The general results show that with no rattles or vibration generated, the helicopter is correctly assessed using A-weighting. With only "a little" rattle, about 10 dB must be added to A-weighted levels for correct assessment, and with "a lot" of rattle, the offset exceeds 20 dB. Overall, these studies demonstrate the very important role noise-induced rattles and vibrations play in noise annoyance.

9:30

**O4. Annoyance from audible infrasound.** Henrik Møller (Institute of Electronic Systems, Aalborg University, Strandvejen 19, DK-9000 Aalborg, Denmark)

For several years it has been known that noise at low and infrasonic frequencies can cause considerable nuisances. The annoyance cannot be predicted by A-weighted sound levels since the A-curve nearly removes the infrasonic frequencies completely and it also seems to attenuate the low audio frequencies too much. Recent work on equal loudness and equal annoyance contours indicates that the annoyance from infrasound is closely related to the loudness sensation. The use of a weighting curve based on the loudness and annoyance curves is suggested. The ISO/DIS 7196 G1 curve might be a proper choice. The applicability of this curve is demonstrated. The experiments involve pure infrasonic tones as well as one-third octave bands. The consequence of simultaneously occurring audio frequency noise is also demonstrated.

### Contributed Papers

9:55

**O5. Broadband rotor noise in atmospheric turbulence.** S. A. L. Glegg (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

This paper describes the development of a prediction method for broadband noise from wind turbines. In this method, two of the more important source mechanisms (unsteady lift and unsteady thickness noise [D. Hawkings, *J. Acoust. Soc. Am. Suppl.* 1 63, S21 (1978)]) can only be predicted if the inflow turbulence intensity and length scale are known. By using a suitable model of the atmospheric boundary layer, these parameters can be estimated, as a function of height above the ground, using the measured wind speed and a ground surface roughness parameter. The sound propagating to the geometric nearfield of the rotor is then calculated by evaluating the source level at 20 azimuthal positions and five radial positions in the rotor disk plane. Other source mechanisms, such as trailing edge noise and the noise from separated flow, are also included. The results are compared with measurements on 5-, 20-, and 80-m-diam wind turbines.

10:10

**O6. Comparison of advanced turboprop and conventional jet and propeller aircraft flyover noise annoyance.** David A. McCurdy (NASA Langley Research Center, Mail Stop 463, Hampton, VA 23665)

A laboratory experiment was conducted to compare the annoyance to advanced turboprop aircraft flyover noise with the annoyance to conventional turboprop and jet aircraft flyover noise. The effects of fundamental frequency and tone-to-broadband noise ratio on advanced turboprop annoyance were also examined. A computer synthesis system was used to generate 18 realistic, time varying simulations of propeller aircraft takeoff noise in which the harmonic content was systematically varied to represent the factorial combinations of six fundamental frequencies ranging from 67.5–292.5 Hz and three tone-to-broadband noise ratios of 0, 15, and 30 dB. These advanced turboprop simulations along with recordings of five conventional turboprop takeoffs and five conventional jet takeoffs were presented at D-weighted sound pressure levels of 70, 80, and 90 dB to 32 subjects in an anechoic chamber. Analyses of the subjects' annoyance judgments compare the three categories of aircraft and examine the effects of the differences in harmonic content among the advanced turboprop noises. The annoyance prediction ability of various noise measurement procedures and corrections is also examined.

10:25

**O7. Reduction of tank main gun noise with a muffler.** Rodney M. Atack, Joel D. Bales, John G. Wrobel, and Nelson D. Lewis (Bio-Acoustics Division, U.S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5422)

With the continuing encroachment of residential communities on military installations and the noise complaints received from these residents, it is necessary to reduce the noise levels propagating from these facilities. At stationary 105-mm tank gun firing locations, such as those used for weapon testing, the noise from the firing can be significantly reduced by firing through a large muffler. The muffler used at one installation to reduce the noise impact is approximately 2.3 m in diameter, 6.3 m long, and weighs approximately 20 metric tons. During a test of this muffler, the noise level in the adjacent community was reduced by approximately 7 dBA from levels that caused a moderate risk of complaints to levels which are barely audible. The significant noise reduction produced through utilizing a noise muffler presents an avenue through which research can be directed towards reducing noise impact from certain types of weapons testing.

10:40

**O8. A noise monitoring and warning system for intermittent large-amplitude impulsive sounds (blasts and sonic booms).** Paul D. Schomer, Aaron Averbuch, and Lester Lendrum (U.S. Army Construction Engineering Research Laboratory, P.O. Box 4005, Champaign, IL 61820)

The noise generated by large weapons (artillery, tanks, demolition) is a source of considerable problems for many Army installations. As a part of the Army's program to mitigate noise problems, the U.S. Army Construction Engineering Research Laboratory (USA-CERL) has designed and installed a blast noise monitoring and warning system at several installations. Each system consists of a base station and several remote, smart sensors. Each smart sensor actually contains two microprocessors and a large memory. When sounds that are "too loud" are detected, the sensor uses a "smart" modem to call the base station computer and warn operations personnel of the high levels. Alternatively, the sensor can buffer (monitor) these data in its memory and read out these data on command (by telephone) to a distant base station. Each sensor, designed for unattended long-term operation, includes such features as self calibration, an uninterruptable power supply, and a  $-40^{\circ}$  to  $+85^{\circ}$  C operating temperature range. Early indications are that the desired reliability and user friendly ease of operation have been attained.

**O9. Identifying prominent discrete tones in noise emissions.** Matthew A. Nobile (IBM Acoustic Laboratory, Department C18, Building 704, P.O. Box 390, Poughkeepsie, NY 12602)

Research has shown that broadband noise containing one or several discrete tones is usually judged more annoying than the same noise without the tones. In view of this, new noise emission standards are requiring that the presence or absence of prominent tones be reported along with the measured data. Therefore, it is imperative that a reliable accurate, and repeatable measurement method be available to identify and evaluate tones in noise. The methods described in current standards are based on the rather outdated, analog procedure that appeared in ANSI S1.13 Appendix A (1971), but as a result of the growing use and affordability of modern digital signal processing instruments, new methods are now being proposed. One method [Nobile *et al.*, *Proceedings of NOISE-CON 85*, Columbus, OH (June 1985)] is based on FFT measurements, while another is based on 1/12-octave-band analysis. This paper will review the concept of a prominent discrete tone, review the proposed new methods, and present comparisons of results from the different methods applied to actual machine noise.

11:10

**O10. Infrasound: Incidence and significance in motor cars.** Sam D. Haddad (Department of Engineering Technology, Western Michigan University, Kalamazoo, MI 49008)

During the last two decades there has been much interest in the effects of infrasound on people. This is partly due to claims made by some scientists that infrasound could cause effects such as intestinal irritations, throbbing in the head, nervous breakdowns, certain allergies, and other unpleasant phenomena of modern life found in industrial cities. Early work [J. E. Green and F. Dunn, *J. Acoust. Soc. Am.* **44**, 144-145 (1968)] concluded a positive correlation between infrasonic thunderstorms and road accidents and other deviant behavior. Others have addressed the question "Does infrasound make drivers drunk?!" With the possible ef-

fects on road safety this paper aims to assess the incidence of infrasonics in transport environments with particular reference to European passenger cars. It was found that infrasonic sound pressure levels (ISPL) were proportional to vehicle speed, window opening, and vehicle size. Also driving with a fitted roof rack tended to increase ISPL by as much as 20 dB. It was also found that under most adverse driving conditions ISPL never exceeded 110 dB which is below human performance thresholds. It seems therefore that the original significance placed on the incidence of infrasound in cars was misplaced. Also it is concluded that further work needs to be done to establish the effects of exposure duration on human performance thresholds and to include the higher low frequencies as well. [Fundamental research project conducted by S. D. Haddad and Charles Marks at Loughborough University, England in 1980 with further investigations by S. D. Haddad concluded in 1983.]

11:25

**O11. A statistical model for predicting the probability of complaints from army weapons noise.** George A. Luz (U.S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5422)

U.S. Army environmental noise assessments use a statistical average (DNL) to evaluate the impact of heavy weapons noise on nearby residents, but experience has shown that variability above the average is the source of complaints. To assist environmental planners and range control officers in predicting complaints from a specific configuration of range and weapon, a BASIC language program short enough for a personal computer has been developed. Users begin by choosing the weapon from a menu and after defining the coordinates of the firing point, target, and observer, they have the option of: (1) entering a noise level and receiving the probability of that level being exceeded at the observer's location, or (2) entering an exceedance level and receiving the noise level associated with that exceedance level. Noise levels are then compared with the criteria used in the successful complaint management program at the Naval Surface Weapons Laboratory, Dahlgren. Onsite blast noise measurements taken over the past decade are used to demonstrate the predictive validity of the model.

WEDNESDAY MORNING, 6 NOVEMBER 1985

SUITE 6, 8:30 TO 12:00 NOON

## Session P. Musical Acoustics II: String Excitation by Plucking, Bowing, or Striking

Donald E. Hall, Chairman

*Physics Department, California State University, 6000 J Street, Sacramento, California 95819*

### Invited Papers

8:30

**P1. Modeling of a piano hammer.** Hideo Suzuki (CBS Technology Center, Stamford, CT 06905)

From the measurement of the force-time pattern and the acceleration-time pattern, a simple model that describes the action of the hammer, when it hits an ideally rigid string, was established. It was found that the relation between the force ( $f$ ) and the deformation ( $x$ ) is approximated by the form  $f = kx^3$  for the specific hammer used for the measurement, where  $k$  is a constant. This shows that the hammer stiffness  $S(x)$  has a nonlinearity of the third order. Then, the effective mass  $M_h$  was calculated using a single mass-spring model. It is fairly constant during the time when the hammer is moving toward the string. The force-time patterns of this model were calculated using  $S(x)$  and  $M_h$  for different hammer velocities. The agreement between the measured and calculated force-time patterns is good (within 15% near their peaks). As a second model, a double mass-spring system was considered. The force-time patterns calculated from this model are in better agreement with the measured curves.

9:00

**P2. Piano string excitation.** Donald E. Hall (Physics Department, California State University, 6000 J Street, Sacramento, CA 95819)

Prediction of piano string motion is challenging for the theorist because the force from the hammer extends over a finite time. Abstract prescriptions for solving this problem with sequences of traveling waves become

much more useful when a microcomputer can be assigned to generate all the proper terms and translate their total into graphical output. Sample results will be discussed for a program that includes hammer compliance and can handle any case where the hammer mass is not too much larger than the string mass. How to make adequate allowance for hammer nonlinearity and finite width, string stiffness, and bridge compliance will be considered, along with possible laboratory measurements that could clarify the application of this theory to piano hammer design.

9:30

**P3. Dynamics of the bowed string.** Robert T. Schumacher (Department of Physics, Carnegie-Mellon University, Pittsburgh, PA 15213)

The bowed string is acted upon by a force, applied at the surface of the string, that is nonlinear in its dependence on string velocity. Consequently, the string's dynamics are affected by rotational motion of the string as well as by the translational motion that largely determines the force that drives the bridge. A review of the theory of this nonlinear bowed string dynamics will be given, followed by a discussion of the effects of rotational motion on starting transients and playing frequency. Experimental results will be shown from a bowing machine, capable of highly reproducible motion, that include measurements of the string's rotational motion during bowing. [Work supported by NSF.]

10:00

**P4. The digital bow.** Gabriel Weinreich (Department of Physics, University of Michigan, Ann Arbor, MI 48109 and IRCAM, 31 rue Saint-Merri, 75004 Paris, France) and René Caussé (IRCAM, 31 rue Saint-Merri, 75004 Paris, France)

The "digital bow" is a device that senses the instantaneous velocity of a chosen point on a string and, by looking up an equivalent frictional characteristic stored in a computer table, exerts an electrodynamic force on the string corresponding to what a physical bow would do under the same circumstances. The process is repeated at a high sampling rate in real time. Under such conditions, we observe the string breaking into spontaneous Helmholtz motion. The arrangement is of great interest to the physicist because it provides exact control of the frictional characteristic of the "bow," allowing one to investigate details of the bowing process to a degree hitherto unattainable. It may also have considerable musical potential, since the behavior of this "bow" is not limited by the properties of Siberian horse hair or of the secretions of dripping evergreens. Video and audio tapes will be presented. [Work supported in part by NSF.]

#### Discussion

10:30-10:45

### Contributed Papers

10:45

**P5. Comparison of new and "dead" nylon guitar strings.** Roger J. Hanson and Gordon O. Munns (Department of Physics, University of Northern Iowa, Cedar Falls, IA 50614)

Classical guitarists are troubled by nylon strings becoming "dead" after a short period (days or weeks) of intensive use. The problem is most severe for the lower three strings which are normally wire wound. New and "dead" strings have been studied under a variety of conditions in an attempt to determine the most evident measurable differences. The time decays of the different partials of the sound radiated from a guitar have been measured, but the interaction of the vibrating string with the guitar body and the difficulty of standardizing plucking and detecting conditions complicate the interpretation. To simplify and standardize the test conditions, the string is mounted on a steel beam with rigid supports. The vibrating motion of the mechanically plucked string is detected optically with a system similar to that used by Gough [C. E. Gough, J. Acoust. Soc. Am. 75, 1770-1776 (1984)]. Measurements of vibration amplitude for different spectral regions as a function of time after the pluck and of the inharmonicity of the partials will be discussed.

11:00

**P6. Jitter in violin and singing-voice tones.** Rudolf A. Rasch (Institute of Musicology, University of Utrecht, Drift 21, 3512 BR Utrecht, The Netherlands)

Musical tones like those of the stringed instruments and the singing voice are not sequences of exactly repeated waveforms with constant frequency, amplitude, and spectrum. Frequency variations can be classified as jitter (random or pseudorandom period-to-period fluctuations), vibrato (quasiperiodic fluctuations with a frequency of 4-7 Hz), and trend (slow,

long-term fluctuations). We have investigated a number of musical tones for the presence of jitter and will present the results of various types of measurements, like period-to-period differences, standard deviations of periods, autocorrelations of periods, etc. One of the main findings was the relation between jitter and the resonance of the sound radiating system (top plate, vocal tract). Tones with frequencies near to a resonance had considerably less jitter than tones with frequencies further removed from a resonance.

11:15

**P7. Analysis of performed transitions in orchestral instruments.** John M. Strawn (The Droid Works/Lucasfilm, P.O. Box CS-8180, San Rafael, CA 94912)

The region between performed notes was examined in nine instruments (flute, bass flute, piccolo, clarinet, oboe, bassoon, trumpet, violin, and cello). On each instrument, eight intervals (2nd, 3rd, 5th, 7th, ascending, and descending) and two playing styles (tongued/untongued, bow change/no bow change) were digitally recorded. Using a computer, the recordings were analyzed for time-varying power and time-varying spectrum (phase vocoder). Analysis of as many as five recordings of a given interval and playing style showed that the performer could easily replicate a given transition, so the recordings were judged as representative. In the tongued (bow change) case, the notes were farther apart, the amplitude dip between the notes (except in the cello) was greater, and spectral changes were more extensive than in the nontongued (no bow change) case. This pattern was not significantly influenced by variation in the size of the interval, the direction of the interval, or the instrument performing. [Work performed at CCRMA, Stanford University, and supported by SDF.]

**P8. A Percussive sensor for playing a computer-controlled synthesizer.** M. V. Mathews and J. Kohut (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A sensor is described for performing a computer-controlled synthesizer. The sensor consists of an equilateral triangle-shaped surface, which is struck with a soft drumstick. Each time the sensor is hit, it sends four pieces of information to the computer—a trigger pulse, the two coordinates of the strike point, and the force of the blow. This information is used by the computer program to control the synthesizer. Several modes of control have been tried. In all modes, loudness is controlled by blow force. In one mode, the two coordinates control pitch and decay time of the sound. Another mode is a form of "conductor program" in which the sequence of pitches to be played is stored in the computer memory. Each stroke causes the next pitch in the sequence to be played. The two coordinates control decay time and spectrum of the sound. The sensor uses three strain gauges mounted at the corners of the triangle. The coordinates and force are computed from the strain gauge outputs by a simple analog circuit.

**P9. Digital recording techniques involving vocal and percussion ensembles: The Gyuto Tantric Monks.** Elizabeth A. Cohen (Department of Physics, Stanford University, Stanford, CA 94305), Michael S. Hart (Grateful Dead; visiting scholar, University of California at Santa Cruz, Santa Cruz, CA 95064), and Huston Smith (Berkeley, CA)

The Gyuto Tantric Monks use handbells, cymbals, damaru (skull drums), rejong (double-membrane wood drums), horns, and long trumpets to summon their deities. Their chants consist of slowly recited Sanskrit texts. They are able to produce and sustain harmonic chords over long periods of time [Smith *et al.*, J. Acoust. Soc. Am. **41**, 5 (1967)]. They produce a music of unparalleled dynamic range. The disparity between the voices and the drums offers a unique challenge to the recording engineer. The Monks were recorded on several occasions in May of 1985. We developed techniques for both live stereo and studio multitrack digital recording. Individual and ambience miking techniques will be discussed. Shared, dynamic, fast-miking techniques for live recordings will be explored. Methods for balancing recording requirements with PA demands will also be discussed. The results of spectral analysis of the recordings confirm the level of fidelity obtained.

WEDNESDAY MORNING, 6 NOVEMBER 1985

DAVIDSON ROOM B, 8:30 TO 11:30 A.M.

### Session Q. Physical Acoustics III: Reflection and Scattering

Laszlo Adler, Chairman

Department of Welding Engineering, Ohio State University, Columbus, Ohio 43210

#### Contributed Papers

8:30

**Q1. Evaluation of beam fields reflected at a plane interface.** Y. Z. Ruan and L. B. Felsen (Department of Electrical Engineering and Computer Science/Microwave Research Institute, Polytechnic Institute of New York, Route 110, Farmingdale, NY 11735)

The complex ray method [L. B. Felsen, Geophys. J. R. Astron. Soc. **79**, 77–88 (1984)] is employed to calculate the fields of a Gaussian beam reflected at a plane interface between two different media. A lateral wave is included in the region of total reflection. To simplify the calculation, a paraxial approximation is employed when the observer is located near the beam axis. These calculations all fail when the beam axis and observer simultaneously approach the critical angle, the criterion of failure being a parameter called the numerical distance. The correct solution in the transition region is generated by a parabolic cylinder transition function. Detailed numerical comparisons provide an assessment of the validity of the various formulations and approximations. [Work supported by NSF.]

8:45

**Q2. Beam excitation of doubly leaky Rayleigh wave on layered viscoelastic half-space.** Finn B. Jensen and Henrik Schmidt (SACLANT ASW Research Centre, 19026 La Spezia, Italy)

The problem of nonspecular reflection of an ultrasonic beam incident at or near the Rayleigh angle onto a thin surface layer overlying a homogeneous viscoelastic half-space is investigated. An approximate solution to this problem was recently published by Nayfeh and Chimenti [J. Acoust. Soc. Am. **75** 1360–1368 (1984)], who presented detailed computational results for a low-velocity surface layer (loaded half-space). We concentrate on the case of a high-velocity surface layer (stiffened half-space), where propagation is characterized by a leaky Rayleigh wave, which, however, will become doubly leaky (energy leaking into the water as well as into the lower half-space) when its phase velocity exceeds the shear velocity of the half-space. For an incident Gaussian beam, we present exact solutions for the reflected field using a recently developed numerical model [H. Schmidt and F. B. Jensen, J. Acoust. Soc. Am. **77**, 813–825 (1985)] providing full wave field solutions for incident beams of arbitrary width, reflected off a multilayered viscoelastic structure. It is shown that

the surface-layer thickness can be determined from the reflectivity pattern for layer thicknesses up to 1.5 times the shear wavelength in the surface layer.

9:00

**Q3. Determination of surface wave velocities on single crystals from backscattering measurements.** Laszlo Adler, Ken Bolland (The Ohio State University, Columbus, OH 43210), Michel deBilly, and Gerard Quentin (Universite Paris 7, 2 place Jussieu, Paris Cedex 05, France)

It was established in previous papers [Adler *et al.*, J. Acoust. Soc. Am. **77**, 1950–1953 (1985)] that the Rayleigh angle backscattering may be used to determine surface wave velocities. The method has been applied to single crystal silver, nickel, and copper. Several planes of symmetry have been considered. Simultaneously, an algorithm has been developed to calculate Rayleigh wave velocities on a single crystal surface. Experimental data agree well with theoretical predictions. [This work was supported by NATO Research Grant #637/84 to The Ohio State University and to the Universite Paris 7.]

9:15

**Q4. Excitation of plate resonances by transient acoustic wave trains.** G. Maze, J. L. Izbicki, J. Ripoché (Université du Havre, Laboratoire d'Electronique et d'Automatique, Groupe "Ultrasons," Uerst, Le Havre 76610, France), A. Nagl, and H. Überall (Department of Physics, Catholic University, Washington, DC 20064)

Experiments have been carried out with 30- $\mu$ s-long sinusoidal wave trains incident on an aluminum plate submerged in water. The excitation of plate resonances by such pulses has been observed, in a fashion used previously for cylinders [G. Maze *et al.*, J. Acoust. Soc. Am. **77**, 1352 (1985)]. The pulse distortions (in the form of initial transients, a quasi steady-state regime, and a final transient) have been interpreted by the interference of reflected and multiple internally refracted pulses, as previously done by us for a layered ocean bottom [A. Nagl *et al.*, Inverse Problems **1**, 99 (1985)]. The final transient, showing a step structure, rep-

resents the ringing of the resonance. Overlapping resonances are shown to lead to beat effects in the ringing. [Work supported by the Direction des Recherches, Etudes et Techniques, France, and by the Office of Naval Research and the Naval Research Laboratory, U.S.]

9:30

**Q5. Transmission and reflection of acoustic waves through a combination of elastic and liquid layers.** Jacob George (Mail Stop 181, Submarine Signal Division, Raytheon, Portsmouth, RI 02871)

We extend the formalism of Folds and Loggins [J. Acoust. Soc. Am. **62**, 1102 (1977)] to calculate acoustic transmission and reflection coefficients for a plane wave at oblique incidence on a combination of elastic and liquid layers of plane parallel plates. We discuss the modified boundary conditions and show how the acoustic field is calculated at any point inside an elastic or liquid layer. We present numerical results for the case of steel and water layers.

9:45

**Q6. Surface wave velocity measurements at fluid-porous solid interface.** Michael J. Mayes, Peter B. Nagy, and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

The presence of ultrasonic surface waves of various modes on a fluid-porous solid interface is demonstrated and their velocities are measured. The experimental technique developed earlier [A. Jungman *et al.*, J. Appl. Phys. **53**, 4672 (1982)] for a fluid-isotropic solid interface utilizes reflected broadband spectra from periodic surfaces. Three sharp minima corresponding to mode coupling of incident waves into surface waves at the fluid-porous solid interface are observed. The velocities of these surface waves are in qualitative agreement with theoretical predictions [S. Feng and D. L. Johnson, J. Acoust. Soc. Am. **74**, 906 (1983)] and are identified as pseudo-Rayleigh, pseudo-Stoney, and true Stoneley waves. [This work is supported by the U. S. Department of Energy through Lawrence Livermore National Laboratory.]

10:00

**Q7. Scattering problems involving fluid-porous interfaces.** C. Thompson and R. Sen (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

The problem of plane-wave scattering by a rigid planar obstacle between a fluid and a fluid-filled porous medium is investigated. Porous materials are used extensively in industry as sound-absorbing liners, and their effect on the incumbent pressure field has traditionally been accounted for by postulating a point-reacting boundary condition at the fluid-porous medium interface. It is shown that such an assumption leads to a considerable misrepresentation of the physical situation. A fluid-porous interface is characterized by a rapid local variation in the mass reactance and flow resistance, and the length scale associated with this variation is fundamental to the scattering analysis. It is shown that local velocity fields at the interface determine important features of the solution, such as the nature of the edge singularity. Pressure and velocity jump conditions at the interface are rigorously derived using the local mass and momentum equations and the method of matched asymptotic expansions. We also compare our result to a solution of the problem that employs the locally reacting boundary condition.

10:15

**Q8. Scattering of ultrasonic waves by cotton fibers.** M. A. Breazeale (Department of Physics, University of Tennessee, Knoxville, TN 37996-1200)

A tone burst of ultrasonic waves in air is scattered on passing through a pad of fibers. The distribution of spatial frequencies within the tone burst is found to be affected by the physical characteristics of the fiber used, and becomes a means of identifying fibers. Known cotton fiber samples whose properties have been studied previously by x-ray diffraction are used to scatter 1-MHz ultrasonic waves in air. Spectrum analysis of

the transmitted tone bursts produces characteristic spectra which correlate well with the x-ray diffraction results. Reliable tests can be made with samples weighing as little as 0.36 g.

10:30

**Q9. Biorthonormalization technique in scattering.** I. Allan G. Dallas, Charles F. Gaumond (Code 5132, Naval Research Laboratory, Washington, DC 20375-5000), and Angie Sarkissian (Sachs/Freeman Associates, Bowie, MD 20715)

The problem of scattering of time-harmonic acoustic waves by a smooth rigid obstacle is considered. A formalism is developed for the construction of the farfield pattern. The method involves the biorthonormalization of two complete families in the Hilbert space of square-integrable functions on the surface of the scatterer. The procedure is more attractive than the orthonormalization technique for the solution of the same problem. [C. F. Gaumond and A. G. Dallas, J. Acoust. Soc. Am. Suppl. **1** **75**, S53 (1984)], since fewer computations are required in applications.

10:45

**Q10. Biorthonormalization technique in acoustic scattering, II.** Angie Sarkissian (Sachs/Freeman Associates, Bowie, MD 20715), Charles F. Gaumond, and Allan G. Dallas (Code 5132, Naval Research Laboratory, Washington, DC 20375-5000)

Results obtained by applying the biorthonormalization method, in the particular case of a rigid prolate ellipsoidal scatterer, are presented for farfield backscattering at axial incidence. Computations based on two forms of the solution are described and compared with the available separation-of-variables calculations [T. B. A. Senior, Can. J. Phys. **44**, 655 (1966)], for the ratios  $a/b = 2, 5$ , and  $10$ , and over the  $ka$  range of  $0-10$  ( $a =$  semimajor axis,  $b =$  semiminor axis, and  $k =$  wavenumber).

11:00

**Q11. Scattering of elastic waves by an anisotropic inclusion.** Vasundara V. Varadan, Jen-Hua Jeng, and Vijay K. Varadan (Laboratory for Acoustic and Electromagnetic Research, Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

Form invariant stress-strain relationships are presented for anisotropic media. The case of transversely isotropic media is then considered and the five elastic constants needed to describe such a medium are obtained in terms of the elements of the stiffness tensor. The dispersion equation is solved and approximate basis functions are constructed to expand the displacement fields inside such an anisotropic inclusion. The scattering of elastic waves by spherical or nonspherical inclusions is studied as a function of frequency and geometry and compared with the isotropic case.

11:15

**Q12. Scattering of ultrasonic pulses by a turbulent water jet.** Murray S. Korman (Department of Physics, U. S. Naval Academy, Annapolis, MD 21402) and Robert D. Corsaro (Physical Acoustics Branch, NRL, Washington, DC 20375)

Experimental results are presented for the scattering of an acoustic pulse from a region of turbulent flow. The investigation was performed in the 7-m-deep section of the USNA Hydrodynamics Tow Tank facility. Turbulent flow is produced by a submerged nozzle of 4.8cm-diam and 100-cm/s exit velocity. Flow is directed perpendicular to the water's surface. The 1.9-MHz acoustic pulse is generated from a 2.54-cm-diam ultrasonic transducer located 18-nozzle diameters downstream from the jet exit and at a perpendicular distance of 84 cm from the jet axis. An identical transducer receiver located at the depth of the sender is positioned 217 cm from the jet axis. Angular scattering measurements are made in a plane parallel with the water's surface. Scattering results are presented for the temporal broadening, the spectral broadening, and the energy distribution of the incident pulse using a digital data collection and analysis system. Results are compared with theoretical predictions. [Work supported by NRL (Physical Acoustics Branch) and USNA.]



## Session R. Speech Communication III: Speech Production I

Gary Weismer, Chairman

*Speech Motor Control Laboratories and Department of Communicative Disorders, University of Wisconsin, Madison, Wisconsin 53705*

## Contributed Papers

8:30

**R1. Spontaneous versus prepared speech.** Jared Bernstein and Gay Baldwin (Speech Research Program, SRI International, Menlo Park, CA 94025)

Some speakers use different forms when training a speech recognizer than when speaking spontaneously to the device—this could be called “enrollment diglossia.” As a preliminary study of this phenomenon, we compared selected phonological and prosodic features of spontaneous speech to read and recited versions of the same sentences and paragraphs. Three subjects were interviewed and were later asked to read or memorize and recite, at various nominal rates, portions of the material that they had originally spoken spontaneously. We made detailed measurements of /t/ allophonics, speech rate, and the forms of certain words. For some speakers, there are considerable differences between spontaneous and prepared renditions; e.g., one speaker produced 45% vs 18% of /t/s as flaps, another speaker varies local speech rate and produces more nonsyntactic pauses in spontaneous paragraphs, and one speaker invokes “fast speech” forms at slower speech rates for spontaneous speech than for prepared speech. [Work supported by DARPA.]

8:45

**R2. Statistical treatment of the acoustic effects of vocal effort.** Kung-Pu Li and Peter Benson (ITT Defense Communication Division, San Diego, CA 92131)

Speech recognition performs best when trained and operated in the same environment. Such training is often impractical, especially in military applications. Apart from the difference in the noise environment, talkers change their speech habits when speaking over poor quality channels. The acoustic changes due to increased vocal effort reduce the performance of speech recognizers. While the acoustic effects of vocal effort are varied, it is possible to deal with a bulk of the differences using a statistical approach. A database of two talkers was collected under conditions which produced a variety of degrees of vocal effort. These conditions included both loud sidetone and noise which, when delivered over headphones, change stress and vocal effort. The talkers differed in their reaction to the extreme level in that one shouted and the other did not. Read materials consisted of lists of digits, randomized but the same across talkers. Acoustical parameters such as pitch, amplitude, and word duration were measured and compared across conditions. Using standard statistical procedures, transforms were computed to map the speech from several conditions into a common transformed space. Recognition experiments demonstrated that significant increases in automatic speech recognition performance can be achieved when templates from one condition are matched against unknown speech, when both have been transformed into the common space. An analysis of the recognition errors was made in light of the acoustic differences between vocal effort conditions.

9:00

**R3. Evaluation of a vowel shadowing task.** Allen A. Montgomery (Army Audiology and Speech Center, Walter Reed Army Medical Center, Washington, DC 20307-5001)

This study was concerned with preliminary steps in the development and evaluation of a shadowing task designed to yield quantitative information on the perceptual-motor processing of speechlike stimuli. In this task, the subject was required to shadow (reproduce vocally as quickly and accurately as possible) a synthetic vowel stimulus (target) which is presented to one ear. The stimuli vary in difficulty from steady-state vowels for calibration, normalization, and training, to continuously varying series of vowels used to test the subject's ability to perform rapid vowel transitions and produce a variety of tense and lax vowel targets. The subject's performance is evaluated by comparing  $F_2$  curves of the produced vowels with those of the targets using cross-correlation and coherence techniques. Results of training and testing ten normal speaking subjects will be presented, and further refinements necessary for developing the tracking procedure into a research/clinical tool will be discussed. [Work supported by the Department of Clinical Investigation, Walter Reed Army Medical Center.]

9:15

**R4. Dynamic and perceptual constraints of loud speech.** Richard Schulman (Department of Linguistics, Stockholm University, 10691 Stockholm, Sweden)

In an investigation of the production of loud speech, it is found that, in comparison with normal speech, these productions are characterized by a substantially lowered jaw position for all vowels. On the basis of these results, we examine whether, for loud speech, speakers maintain tongue constrictions typical for normal vowel production in spite of the lowered jaw position, or if the tongue follows the jaw using constrictions atypical for the particular vowel. Acoustic analyses of the loud tokens show an increase in  $F_1$ , thus indicating a lowering of the tongue. Tests show, however, that listeners perceive the intended vowel quality, and not that which might be inferred from the loud formant values. The characteristics of the loud speech targets will be discussed in light of the dynamic constraints of the articulatory system and the perceptual constraints set by the listener.

9:30

**R5. Acoustic-phonetic properties of vowels produced in noise.** David B. Pisoni and Moshe Yuchtman (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

It is well known that speech produced in noise has higher intensity, longer duration, and higher fundamental frequency than speech produced in quiet. However, recent studies in our laboratory have indicated that changes in the acoustic properties of speech produced in noise are not limited to changes in the glottal source but may also include significant changes in the acoustic-phonetic structure of speech reflected in the characteristics of vocal tract transfer function. We investigated the effects of wideband noise on the acoustic properties of vowels. Talkers produced ten American-English vowels in /hvd/ context. Ten tokens of each vowel were produced in quiet and in the presence of masking noise (90 dB SPL). The results indicate a rather complex pattern of changes in formant frequencies. First, with the exception of the vowels /æ, a, ʌ/, whenever a significant change in  $F_1$  was observed,  $F_1$  frequency was higher in the noise condition. Second,  $F_2$  frequencies decreased in front vowels and increased

in back vowels. The net results of these changes is a more compact vowel space.

9:45

**R6. Labial articulation patterns associated with segmental features and syllable structure in English.** Marian J. Macchi (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

Linguistic contrasts in syllable structure were compared with contrasts in segmental structure by studying lower lip movement data obtained with the University of Tokyo x-ray microbeam system [Kiritani *et al.*, J. Acoust. Soc. Am. 57, 1516–1520 (1975)]. Intervocalic [p] and [s] in three syllabic structures (V\$CV, VC\$V, VC\$CV) with different vowel environments (combinations of [i], [a], [u]) were analyzed for two speakers. More complex structures (V\$[p]V, V[p]\$[l]V, V[p]\$V, V\$[sp]V, V[s]\$[p]V, V\$[sp]V, V[s]\$[p]V) were also analyzed for one of the speakers. Lower lip height was separated into jaw and lip-proper components [Edwards *et al.*, J. Acoust. Soc. Am. Suppl. 1 74, S117 (1983)]. Several relationships between jaw height and lip-proper height were observed. Segmental contrasts (e.g., [p] vs [s]; [p] in [ipi] vs [apa]) were realized by height differences in both the jaw and the lip proper, whereas syllable structure contrasts (e.g., syllable-initial [p] vs syllable-final [p]) were realized primarily by jaw height differences. Thus the phonological distinction between segmental and suprasegmental contrasts was maintained at the level of articulatory representation.

10:00

**R7. A dynamic analysis of reiterant speech production in Japanese.** Eric V. Bateson and J. A. S. Kelso (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Previously, we have modeled supralaryngeal articulatory motion in the production of English reiterant speech in terms of underlying dynamical constraints on movement gestures [Kelso *et al.*, J. Acoust. Soc. Am. 77, 266–280 (1985)]. Here, we extend the model to Japanese, a language whose temporal structure is believed to be based on the “mora” and thus very different from English. Preliminary analysis of acoustic lip and jaw movement, and electroglottographic data from three Japanese talkers reveal production differences between the reiterant renditions of multimora and single mora syllables. Multimora CVC syllables are characterized by a glottally truncated vowel followed by a lengthened silent period prior to voicing onset for the following syllable, e.g., a word like (motte) is realized reiterantly as [ba?ba]. Thus the timing of the glottal gesture appears specific to the mora structure of Japanese, while upper articulator motion, which does not differ for the two syllable types, can be shown to adhere to dynamic constraints hypothesized for movement gestures in general. [Work supported by NINCDS and ONR.]

10:15

**R8. Phase-dependent sensitivity to perturbation reveals the nature of speech coordinative structures.** K. G. Munhall and J. A. S. Kelso (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The detailed workings of hypothesized speech coordinative structures—functionally specific articulator ensembles—can be explored by varying the phase of an unexpected perturbation during ongoing speech. The present research examined the characteristics of remote (nonmechanically linked) and autogenic compensation when the lower lip was perturbed in either the opening or closing phase of the motion for the utterance /baeb/. Preliminary results indicate that the relative contribution of remote and autogenic responses to the perturbation varies with the phase of perturbation; remote compensation plays a greater role when the lower lip is perturbed during the closing phase of motion for a bilabial consonant. Analysis of trials immediately following perturbed trials reveals a hysteresis in the articulator configuration due to the preceding loaded trials. The articulators tend to overcompensate in these trials in a manner that reflects the phase sensitivity of the immediate compensation pattern. These phase-sensitive compensation patterns provide strong support for the notion that the articulators in speech are flexibly assembled as coor-

dinative structures for accomplishing specific phonetic goals. [Work supported by NINCDS and NSERC.]

10:30

**R9. Articulatory movements—Icebergs revisited.** O. Fujimura (AT&T Bell Laboratories, Murray Hill, NJ 07974)

In previous meetings, we discussed a hypothesis that there are relatively robust parts of CV and VC (or, more generally, any demissyllabic) movement patterns for the peripheral articulators, that are directly relevant to the place-specifying consonants involved. This paper attempts to clarify the limitation of the phonetic environments in which such an “iceberg” invariance is maintained, to define the crucial and robust articulatory dimensions from this point of view, and to examine the compatibility of this hypothesis with other models of articulatory movements. It is shown that extremely fast (probably inexact) utterances can introduce a remarkable difference in movement patterns, particularly in repeated utterances, whereas rather extreme contrastive emphasis conditions that considerably alter vowel gestures do not. A comparison between isolated words and sentence-embedded words seems to support the invariance notion.

10:45

**R10. Articulatory contact patterns in /t/ and /s/ produced with and without a bite block.** J. E. Flege and S. G. Fletcher (Biocommunication, University of Alabama, Birmingham, AL 35294)

Dynamic palatometry was used to examine linguapalatal contact during consonant production. The subjects (two native English, three native Arabic) produced /s/ and /t/ in two normal speech samples immediately after insertion of a bite block and, again, after 10 min of practice with the bite block. One English subject contacted more sensors when producing /s/ with the bite block, the other contacted fewer sensors. Two Arabic subjects showed an increase in the width of the /s/ groove, along with a decrease in the number of sensors contacted; the other Arabic subject showed a decrease in groove width and an increase in the number of sensors contacted. Only one of the Arabic subjects showed significant adaptation, producing /s/ with a narrower (more normal) groove width in bb2 than bb1. Inserting the bite block did not affect the width of A-P constriction used by the English subjects to form /t/ but did lead to an increase in the number of sensors contacted. On the other hand, all three Arabic subjects contacted fewer sensors when producing /t/ with a bite block. The Arabic subjects all formed a /t/ with narrower constriction in the bite-block samples, but only two showed significant adaptation from bb1 to bb2. The intersubject differences will be discussed in terms of cross-language phonetic and individual morphological differences.

11:00

**R11. On velopharyngeal articulation in the origin of human CV contrasts.** Harold R. Bauer (Speech and Hearing Science Section, The Ohio State University, 324 Derby Hall, 154 N. Oval Mall, Columbus, OH 43210)

The origin of human CV contrasts from nonhuman primate sound systems is hypothesized to be, in part, dependent upon effective nasal port valving due to velopharyngeal articulation. Common chimpanzee sound acoustics and associated oral-facial movements were studied. Compared with obstruents found in human infants, chimpanzee sounds suggest no similar post-velum evidence of obstruent production. An ethologic model of phonetic development based upon the production of sensorimotor contrasts [Bauer, J. Acoust. Soc. Am. Suppl. 1 75, S45 (1984)] was used to detail the hypothetical phylogenetic and developmental significance of nasal port articulation in the origin of human speech sound contrasts. Consistent with the hypothesis, the lack of comparative evidence for chimpanzee post-velum obstruent sounds supports the significance of velopharyngeal articulation in the origin of human obstruent-vowel contrasts.

11:15

**R12. The new nationally shared x-ray microbeam facility: Status report.** R. Nadler, J. H. Abbs, M. Thompson (University of Wisconsin,

A 600-kV x-ray microbeam system for studying tongue movements and other articulatory gestures has been constructed to serve as the core instrument of a nationally shared speed production research facility at the Waisman Center, University of Wisconsin. Preliminary speech movement data have been obtained and this system currently is capable of tracking multiple articulatory pellets (up to 12) at aggregate sampling

rates of about 1000 per second and simultaneous A/D sampling of multiple EMG and acoustic signals. The facility includes parallel capability for data display and analysis for multiple experimenters. It is expected that this system will be used by many U. S. research groups. The administrative structure, as well as technical specifications of the facility will be described, and proposals for its use will be invited. [Support for this facility is provided by NINCDS (NS-16373).]

WEDNESDAY MORNING, 6 NOVEMBER 1985

DAVIDSON ROOM A, 9:00 TO 11:30 A.M.

## Session S. Physical Acoustics IV: Nonlinear Acoustics

Joseph E. Blue, Chairman

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

### Contributed Papers

9:00

#### S1. A coefficient of nonlinearity for noncollinear plane-wave interaction.

Mark F. Hamilton and James A. TenCate (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

An inhomogeneous wave equation, exact to second order in the field variables, is derived for the sum and difference frequency pressure generated by two plane waves, of angular frequencies  $\omega_1$  and  $\omega_2$ , which intersect at an angle  $\theta$  in a lossless fluid. The coefficient of nonlinearity pertaining to the sum or difference frequency wave ( $\omega_{\pm} = \omega_1 \pm \omega_2$ ) is shown to be

$$\beta_{\pm}(\theta) = B/2A + \cos \theta \pm 4(\omega_1/\omega_{\pm})^2 \sin^4(\theta/2),$$

where  $B/A$  is the parameter of nonlinearity. The same result may be deduced from the work of Zverev and Kalachev [Sov. Phys. Acoust. **15**, 322 (1970)]. The first term of  $\beta_{\pm}$  is due to the isentropic equation of state, the second term represents convection, and the third term comes from the momentum equation. Alternative formulations of the inhomogeneous wave equation are presented, and comparisons are made with the analyses of others. An experiment designed to measure the angular dependence of  $\beta_{\pm}$  was conducted with noncollinear waves in an airfilled waveguide. Results are reported. [Work supported by ONR.]

9:15

#### S2. Wave steepening in Van der Waals gases. M. S. Cramer and R. Sen (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

Van der Waals gases having sufficiently large specific heats are known to possess regions of negative nonlinearity, i.e., regions where the nonlinearity parameter  $1 + B/2A < 0$ . When the amplitude of a pulse or wavetrain is sufficiently large, different parts of the wave may correspond to positive and negative values of the nonlinearity parameter, and the resultant propagation will be qualitatively different than that observed in ideal gases. A complete finite amplitude theory of the steepening of both pulses and wavetrains will be presented. Although monosigned pulses in ideal gases and many liquids form no more than one shock wave, it will be shown that, under the same conditions, Van der Waals gases may form as many as three shock waves.

9:30

#### S3. Fourier series representation of finite amplitude sound beams. Hsu-Chiang Miao and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

An earlier analysis of finite amplitude sound beams derived general expressions from the behavior in the off-axis region [J. H. Ginsberg, J. Acoust. Soc. Am. **76**, 1201-1214 (1984)]. An asymptotic analysis of the region very close to the beam axis confirms the earlier result for the velocity potential. The signal is rewritten in a form that makes the contribution of each wavenumber in a continuous spectrum appear to be the sum of two waves traveling transversely, as well as axially. The coordinate transformations required to renormalize this form lead to a temporal Fourier series that is reminiscent of the Fubini solution for finite amplitude planar waves. The complex amplitude of each harmonic is obtained from an integration over the transverse wavenumber. The computational efficiency of this representation permits extensive evaluation of propagation properties. An example compares the signal derived from a piston to that obtained from the one-dimensional assumption that  $p = \rho c v_n$  on the boundary, which has been employed in prior investigations using approximate parabolic equations. [Work supported by ONR, Code 425-UA.]

9:45

#### S4. Finite amplitude acoustic waves generated by a baffled dual frequency transducer. Mosaad A. Foda and Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

In a previous presentation [J. Acoust. Soc. Am. Suppl. **1** **75**, S92 (1984)], a nonuniformly accurate expression for the velocity potential was derived using a singular perturbation procedure. The linearized signal was represented by a dual King integral, and the cumulative growth was evaluated asymptotically. In this paper, the renormalization version of the method of strained coordinates is employed to annihilate the secular terms. The result is a uniformly valid expression for the acoustic pressure at all locations preceding the formation of a shock. Additional discussion is devoted to a simpler model that interfaces the dual frequency King integral in the nearfield with nonlinear spherical distortion in the farfield. The results obtained from both descriptions in the case of a parametric array (proximate primary frequencies) agree better than earlier theories with previous measurements performed for a wide range of parameters. [Work supported by ONR, Code 425-UA.]

10:00

#### S5. Theoretical description of a focused Gaussian beam. Gonghuan Du and M. A. Breazeale (Department of Physics, University of Tennessee, Knoxville, TN 37996-1200)

The nonlinear equation in nondimensional form used by S. I. Aanonsen *et al.* [J. Acoust. Soc. Am. **75**, 749 (1984)] is solved to obtain an expression for the sound-pressure distribution produced by a plane Gaussian radiator with a concave solid lens in an absorbing fluid medium. Analysis of the solutions shows that the sound field of a focusing lens still maintains a Gaussian distribution across the beam without maxima and minima in the nearfield as well as without sidelobes in the farfield. The theory also shows that the focal distance is not equal to that predicted by geometrical acoustics, but is always shorter because of diffraction. The second harmonic pressure also exhibits Gaussian behavior throughout the beam, even in the focal plane, where the second harmonic is observed to be more narrow than the fundamental. The growth of total energy in the second harmonic also is evaluated. The results indicate that the second harmonic initially increases quite rapidly up to the focus. Beyond the focus, the rate of growth of the second harmonic is found to decrease, and, at a certain distance, the second harmonic is found to have an even smaller magnitude than would be the case without focusing. A set of curves is presented which helps to describe the features of a focused Gaussian beam. [Research supported by the Office of Naval Research.]

10:15

**S6. Noncollinear interaction of two sound beams from displaced Gaussian sources.** Mark F. Hamilton, Jacqueline Naze Tjøtta,<sup>a)</sup> and Sigve Tjøtta<sup>a)</sup> (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

We consider the noncollinear interaction of two sound beams in a nondissipative fluid, one formed by a high-frequency (pump) wave and the other by a low-frequency (source) wave. It is assumed that both waves are generated by Gaussian shaded sources. However, the sources may possess linear phase shading, different spot sizes, and their centers may be displaced by a small distance relative to each other. The analysis is based on the fundamental theory developed in earlier papers [G. S. Garrett *et al.*, J. Acoust. Soc. Am. **75**, 769 (1984) and M. F. Hamilton, J. Acoust. Soc. Am. **76**, 1493 (1984)], and it is valid throughout the paraxial field. Numerical results are presented, and asymptotic formulas which explain the development of the parametric field along the axis of the pump wave are given. The analysis may be applied to a parametric receiving array located in the nearfield of the sound source. [Work supported by ARL: UT IR&D funds.] <sup>a)</sup> On leave from Department of Mathematics, The University of Bergen, 5000 Bergen, Norway.

10:30

**S7. Nonlinear viscous effects arising near a boundary discontinuity.** Charles Thompson (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

The problem of nonlinear viscous flow generated by an acoustic wave traveling in proximity to a rigid boundary having a rapid variation in shape will be examined. Special attention will be given to the problems of acoustic streaming and instability in the Stokes boundary layer. The acoustic particle velocity amplitude for which instability is observed corresponds to the case where  $\epsilon^2 R^{1/2}/S^2 = O(1)$ . Below this amplitude threshold, acoustic waves generate time-average motion which is called acoustic streaming. Preliminary results show that the boundary geometry plays a major part in the instability mechanism in that an inflection point in the wall geometry gives rise to a bifurcation point according to linear theory. This state of affairs is in marked contrast to Stokes layers generated over flat boundaries, which exhibit no such bifurcation point. By virtue of the compressible character of the base flow, the characteristic acoustic scale (namely, the acoustic wavelength) does influence the spatial development for the vortical disturbance field. Therefore, an analysis of the relative influence of geometric and acoustic length scales in the development of the distribution field will also be presented.

10:45

**S8. Measurements of the nonlinear tuning curves of Helmholtz resonators.** Junru Wu and Isadore Rudnick (Department of Physics, University of California, Los Angeles, CA 90024)

The tuning curves of several Helmholtz resonators terminating impedance tubes have been measured for sound-pressure levels at the necks of the resonators of from 18.8 dyn/cm<sup>2</sup> to  $1.36 \times 10^4$  dyn/cm<sup>2</sup>. It is found that the peak frequencies of the tuning curves shift to higher frequencies as the sound level is increased. The shifting is believed to be due to the fact that the end correction of a Helmholtz resonator is a function of the sound level at the neck; the higher the sound level, the smaller the end correction [U. Ingard, J. Acoust. Soc. Am. **25**, 1037 (1953)]. A peak frequency shift as high as 20% has been observed. [Work supported by ONR.]

11:00

**S9. Excitation of underwater wall-flex oscillation under static conditions.** S. A. Elder (Physics Department, U. S. Naval Academy, Annapolis, MD 21402)

In a previous paper, resonant acoustical oscillations of a free-flooding cavity were described. These take place when the mouth of the cavity is located along the side of a towed model with flexible walls. It has been the working hypothesis that oscillation amplitude for this case is limited by the nonlinear acoustical impedance of the cavity mouth, in a fashion analogous to self-excited Helmholtz resonance in air cavities. In order to investigate the nonlinear response characteristic of the cavity mouth, tests have been made, under static conditions, using an array of J9 projectors to stimulate wall vibration. Detailed mapping of wall motions has been performed by means of accelerometers. Although the complexity of the wall vibrational modes makes it difficult to reproduce the exact conditions of towed-model cavity tones, it has been possible to associate large fluctuations of internal sound pressure with certain normal modes that periodically alter the cavity cross section. The search for nonlinear mouth flow effects has been complicated by the presence of amplitude-dependent vibrational states of the walls. At this writing, it appears that the present feedback model of wall-flex oscillation may have to be modified to allow for nonlinear wall behavior.

11:15

**S10. Nonlinear evolution of three-dimensional vortical disturbances in a slowly varying waveguide.** K. Herbert, S.-L. Kuo, and C. Thompson (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

The instability of the Stokes boundary layer generated by an acoustic disturbance in a two-dimensional waveguide having a slowly varying height is examined. As the disturbance amplitude becomes finite in value, it is found by Thompson [J. Acoust. Soc. Am. Suppl. **1** **77**, S48 (1985)] that the Reynolds stress causes the solution for the disturbance field to bifurcate supercritically from that obtained by linear stability theory. Therefore, special attention is given to the problem of weakly nonlinear evolution of 3-D vortical disturbances. The disturbance amplitude is shown to be governed by the solution of the nonlinear Schrödinger equation

$$\frac{\partial^2 A(x, \tau)}{\partial x^2} + (\gamma_1 x + T_1 \gamma_2) A + \gamma_3 A |A|^2 = \gamma_4 \frac{\partial A}{\partial \tau},$$

where  $A$  is the disturbance amplitude, and  $\gamma_1$ ,  $\gamma_2$ ,  $\gamma_3$ , and  $T_1$  are parameters dependent on the local behavior of the acoustic wave, as well as the duct geometry. The solution for time-spatial evolution as well as the global behavior of the disturbance field, will be presented.

## Session T. Physiological and Psychological Acoustics II: Performance of Impaired Listeners

Igor V. Nabelek, Chairman

Audiology and Speech Pathology Departments, University of Tennessee, Knoxville, Tennessee 37996

## Contributed Papers

9:00

**T1. Hearing-aid microphone systems with increased directionality.** William M. Rabinowitz, Daniel A. Frost, and Patrick M. Peterson (Research Laboratory of Electronics, Room 36-761, MIT, Cambridge, MA 02139)

Under some conditions of background noise and/or reverberation, increasing the directionality of hearing-aid microphone systems can lead to gains in speech intelligibility. Conceptually, directional microphone systems are realized by combining the outputs of multiple, spatially distributed omnidirectional sensors. Two approaches to obtaining increased directivity  $D$  derive from the classical antenna theory: (1) With uniform or tapered sensor weightings, increasing the spatial extent  $L$  of the sensor array increases  $D$ . The maximum  $D$  is limited to  $2Lf/c$ , where  $f$  is frequency and  $c$  is the speed of sound. (2) With arbitrarily small arrays, broadband "superdirectivity" can be achieved whereby the maximum  $D$  is independent of frequency and equals the square of the number of sensors in the array. A conventional hearing-aid directional microphone exploits suboptimum superdirectivity; two closely spaced sensors attain  $D = 4.5$  dB up to about 4 kHz. We shall review the basic principles and limitations of such microphone systems and report on our effort to develop a four-sensor array (using two conventional directional microphones) that achieves  $D = 8.5$  dB and a 3-dB beamwidth of  $\pm 37^\circ$  for  $f < 4$  kHz with an array length  $L = 4.1$  cm that is practical for hearing-aid applications. [Work supported by NIH.]

9:15

**T2. Spectral profile analysis in the hearing impaired.** F. L. Wightman, M. E. Perkins, and P. Allen (Project Phoenix, Inc., Madison, WI 53705)

We tested the hypothesis that hearing-impaired listeners with reduced frequency selectivity have abnormal difficulty discriminating spectral patterns, especially when degraded by background noise. Both normal and hearing-impaired listeners were asked to discriminate a stimulus with a flat spectrum from one with a rippled spectrum. A two-interval forced-choice paradigm was used, in which overall stimulus level was randomized from interval to interval. The stimuli consisted of random-phase sinusoidal components spaced at half-normal critical band intervals. The spectral ripples were sinusoidal on dB versus critical-band coordinates. The depth of the ripple was fixed (e.g., at 6 dB or at 10 dB) for a block of trials, and a wide-band Gaussian noise was added, with its level varied adaptively from trial to trial to track the signal-to-noise ratio required for 71% correct discrimination. Normally hearing listeners show large individual differences, both in discrimination and in duration of practice effects. However, the effect of noise was the same across all normal listeners: once the stimuli were a few (5 or less) dB above masked threshold, discrimination was asymptotic. Hearing-impaired listeners have much greater difficulty discriminating spectral profiles, particularly in a noise background.

9:30

**T3. Critical bands in the perception of speech signals. Part 2: SNHL data.** Gordon R. Bienvenue (State University of New York, New Paltz, NY 12561) and Robert D. Celmer (College of Engineering, University of Hartford, West Hartford, CT 06117)

The critical band is essentially a filtering process of the auditory system whereby sound is categorized according to its frequency content. While some empirical evidence suggests that the critical band may serve in the analysis of speech, this has only been indirectly demonstrated. The authors have developed digital signal processing techniques for altering available recorded speech materials so that the frequency resolution may be controlled. Tapes have been produced wherein the frequency resolution is limited to narrower than, equal to, and wider than one critical band. In a previous paper [Celmer *et al.*, J. Acoust. Soc. Am. Suppl. 1 76, S14 (1984)], it was reported that these tapes were used in intelligibility testing with normal listeners. Evidence of distorted critical bands in some sensorineural hearing impaired listeners has been reported in the literature. In the present study, the tapes described above were used in intelligibility testing of listeners with confirmed cases of sensorineural hearing loss. The critical bandwidths of the subjects were also measured by a two-tone masking technique, and were shown to correlate highly with the results of the bandwidth-resolution-limited speech test.

9:45

**T4. Measurement of the  $2f_1$ - $f_2$  combination tone in a subject with a noise-induced hearing loss.** Gail A. Takahashi and M. Jane Collins (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

A cancellation technique was used to measure the  $2f_1$ - $f_2$  combination tone in a subject with a bilateral asymmetric noise-induced hearing loss. The study was designed to determine the effects of primary frequency and level on the perception of the  $2f_1$ - $f_2$  combination tone. The  $f_2/f_1$  ratio was 1.2, and six primary tone pairs were chosen such that their frequencies fell below, in, and above the frequency region of the notched hearing loss. In the subject's poorer ear, the  $2f_1$ - $f_2$  combination tone was not perceived when the combination tone frequency fell in the region of the hearing loss. When the primary frequencies fell in the region of the hearing loss, the  $2f_1$ - $f_2$  combination tone was not perceived for input levels of 50 and 60 dB SPL but was for 70 dB SPL.

10:00

**T5. Use of time-compressed speech in the assessment of central nervous system disorders.** Jane A. Baran (Department of Communication Disorders, University of Massachusetts, Amherst, MA 01003 and Dartmouth-Hitchcock Medical Center, Hanover, NH 03756), Susan Verkest, Karen Gollegly, Karen Kibbe-Michal (Dartmouth-Hitchcock Medical Center, Hanover, NH 03756), William F. Rintelmann (Wayne State University School of Medicine, Detroit, MI 48200), and Frank E. Musiek (Dartmouth-Hitchcock Medical Center, Hanover, NH 03756)

Twenty-seven subjects with surgically, radiologically, or neurologically confirmed lesions of the central nervous system were administered a 60% time-compressed version of the NU-6 word lists. Subjects ranged in age from 12 to 59 years. Twenty-four subjects had normal hearing (25 dB HL or better) bilaterally at 500 to 4000 Hz. Three subjects demonstrated a mild hearing loss at a single frequency in one ear. Test stimuli were presented at 40 dB SL *re*: speech reception thresholds. Percent correct scores were derived for each ear and compared to norms previously established [D. Beasley *et al.*, J. Aud. Res. 12, 71-75 (1972)]. Results revealed

that in 67% of the subjects tested, performance in at least one ear fell below established norms. For subjects with abnormal thresholds, performance was abnormal in the "better" ear, or in both ears, in all three cases. These results suggest that a time-compressed speech test may be moderately useful in the identification of CNS lesions. The implications of adding this test to a neuroaudiological test battery will be discussed.

10:15

**T6. Recognition of dichotically presented rhyme words in split-brain patients.** F. E. Musiek, K. Gollegly, K. Kibbe-Michal, and S. Verkest (Department of Audiology, Dartmouth-Hitchcock Medical Center, 2 Maynard Street, Hanover, NH 03755)

A dichotic rhyme word task was administered to six young adult, right-handed, split-brain subjects with normal peripheral hearing. Two of these subjects were tested before and after surgical sectioning of the corpus callosum while the remaining four were tested post-operatively only. The dichotic task is composed of synthetic, well-aligned, rhyme words which were presented at 50 dB SL (*re*: spondee thresholds). The task is such that normal listeners repeat either the word presented to the right or left ear but seldom repeat both simultaneously presented stimuli. Therefore, normal performance is approximately 50% for each ear with a slight right-ear advantage. The split-brain subjects performed uniquely on the dichotic rhyme task. As expected, there was a severe left-ear deficit for verbal report of the stimuli. However, there also was a marked right-ear enhancement on this dichotic task. These trends were not present for the two subjects tested prior to surgery indicating that callosal sectioning was responsible for this phenomenon. The right-ear enhancement noted on this dichotic task may be an indication of "release from central auditory competition" for the left hemisphere.

10:30

**T7. Brainstem auditory evoked potentials (BAEPs) and audiograms after severe closed head injury (CHI).** J. Braxton Suffield and Ken B. Campbell (Division of Neuropsychology (K-11), Henry Ford Hospital, Detroit, MI 48202 and School of Psychology, Ottawa University, Ottawa, Ontario, Canada K1N 6N5)

The use of the BAEP has largely been restricted to prognosticating from acute states, with few long-term studies after injury. We tested both ears of 20 survivors of severe CHI ( $\bar{X}$  coma duration = 28 days) on average 3.2 years after injury. Replicate averages of artifact-free BAEPs were recorded from Cz (passband 100–3000 Hz), with an ipsilateral low mastoid reference. A cutoff of 2.5 standard deviations from lab norms was used in discriminating normal from abnormal responses. In general, few patients had significantly abnormal BAEPs or audiograms. Although seven of the 22 patients had delays in the latency of peak I in one or both ears, all but one of these patients had bilaterally normal IPLs. Nonparametric correlations indicated left and right audiograms were correlated ( $p < 0.01$ ), as were results from left and right BAEPs ( $p < 0.02$ ). BAEPs and audiograms were also significantly correlated ( $p < 0.01$ ). These results tend to validate the concept that lower structures, including those in the brainstem, suffer last and least, even in cases of very severe head injury.

10:45

**T8. Encoding voice pitch for profoundly hearing-impaired listeners.** Ken W. Grant (Central Institute for the Deaf, Saint Louis, MO 63110)

The ability of five profoundly hearing-impaired subjects to "track" connected speech and to make judgments about the intonation and stress in spoken sentences, was evaluated under a variety of audiovisual conditions. These included speechreading alone, speechreading plus speech (low-pass filtered at 4 kHz), and speechreading plus a tone whose frequency, intensity, and temporal characteristics were matched to the speaker's fundamental frequency ( $F_0$ ). In addition, several frequency transfer functions were applied to the normal  $F_0$  range resulting in new ranges that were both transposed and expanded with respect to the original  $F_0$  range. Three of the five subjects were able to use any of the tonal representations of  $F_0$  nearly as well as speech to improve their speechreading rates and to make appropriate judgments concerning sentence intonation and stress. The remaining two subjects had difficulty integrating visual and auditory information despite intensive training periods lasting over a year, even though they were able to make appropriate judgments concerning intonation and stress with expanded  $F_0$  signals when not speechreading. [Work supported by a grant from NINCDS (NS 03856) to Central Institute for the Deaf, St. Louis, MO.]

11:00

**T9. Residual hearing in deafness.** Anne-Marie Hurteau and Michel Picard (Department d'orthophonie et audiologie, Université de Montréal, Montréal, Québec, Canada H3T 1A8)

The purpose of the investigation was to determine whether the nature of audiometric responses obtained from listeners with profound hearing loss are auditory or vibro-tactile. Two types of coupling systems were used to measure the sensitivity thresholds of seven adults with corner audiograms. One coupling system consisted of a standard TDH-39 earphone mounted on a MX41-AR cushion. In the other coupling system, a speculum was added to the earphone cushion in order to reduce the area of acoustical stimulation. Sensitivity thresholds and temporal integration functions were determined at 125, 250, and 500 Hz. The results obtained from individual subjects made it possible to clearly differentiate between auditory and vibro-tactile responses. The findings of the investigation will be discussed in terms of the potential perceptual abilities of listeners with profound sensorineural hearing loss.

11:15

**T10. Contralateral overmasking effects on Békésy audiometry.** I. M. Young and L. D. Lowry (Department of Otolaryngology, Thomas Jefferson University, Philadelphia, PA 19107)

Threshold and amplitude measurements of fixed frequency Békésy tracings were made both in the quiet and in the presence of contralateral white noise masking with 100 dB and 120 dB SPL for 28 subjects with unilateral sensorineural hearing impairment. Masking resulted in one of the following effects on the threshold and amplitude of the test ear: (1) no change in threshold and amplitude for both interrupted and continuous tones, (2) threshold shift for both interrupted and continuous tones by increased masking level with no change in amplitude, (3) threshold shift for both interrupted and continuous tones with increasing separation and reduction in amplitude of continuous tone by increased masking level. Factors affecting threshold and amplitude were discussed with relation to cross masking, central masking, adaptation and pathological conditions of hearing impairment, and were compared with effects of ipsilateral masking.

## Session U. Noise III: Noise Generation and Analysis

David K. Holger, Chairman

*Department of Engineering Science and Mechanics, Iowa State University, Ames, Iowa 50011**Contributed Papers*

1:00

**U1. Acoustic wave propagation in a radial duct.** David K. Holger (Department of Engineering Science and Mechanics and the Engineering Research Institute, 214 E.R.I. Bldg., Iowa State University, Ames, IA 50011) and Mike Wakumoto (Department of Freshman Engineering, Iowa State University, Ames, IA 50011)

The results of an experimental investigation of acoustic wave propagation through a radial duct are reported. The investigation was limited to large wavenumbers in the absence of flow, and experimental results are compared to a corresponding analytical model. The apparatus consisted of an anechoic termination, one rigid duct wall, and one variable impedance duct wall. Preliminary experiments revealed that the measurement techniques were appropriate and that the apparatus was behaving as expected. Experimental plots of pressure versus radial position were found to be in agreement with analytical predictions for cases involving two rigid duct walls. A variable backing depth Helmholtz resonator array was substituted for one of the rigid duct walls, and measurements of pressure versus radial position were made for three backing depths. Values of radial attenuation were estimated from the pressure measurements using a large wavenumber approximation. Experimental radial attenuation estimates are in good qualitative agreement with analytical predictions.

1:15

**U2. Excitation of piping systems from fully turbulent internal flow.** J. M. Cuschieri (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

A source of excitation in pipe systems is the internal, fully developed turbulent flow. This excitation is dependent on the roughness and speed of flow and the response of the pipe structure is very much controlled by the coupling between the structure and the internal medium. This coupling is very strong if the internal medium is a liquid such as water, while it is rather weak but not negligible if the internal fluid is very light such as air. In the past, the treatment of this excitation problem was through the use of joint acceptance functions. The excitation is random in nature and the pipe will accept vibrational energy preferably at its resonant modes. At high frequencies the modal density is high and thus the pipe accepts energy almost at any frequency. Using the joint acceptance function will not give the physical insight required, however, some assumption can be made regarding the joint acceptance function which will then allow the development of an expression which gives a physical understanding of the excitation mechanism. Using this approach, phenomena such as the decrease in the transmission loss of the pipe wall with increasing flow speed, the relation between the turbulent pressures and the excitation forces, and the structural response that can be used to determine the level of vibrational energy input into the pipe structure, can be deduced.

1:30

**U3. The development of an anechoic termination for thin walled shell structure.** R. F. Schapley, II, J. M. Cuschieri, and S. E. Dunn (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

In a series of experiments on the flow of vibrational energy in pipe structures and on the coupling between the structure and the internal medium during the vibration of the outer shell, it was desired that a struc-

ture with similar characteristics to that of an infinite structure be developed. That is, a pipe structure with anechoic terminations was required. It was decided that the best approach would be to embed the pipe structure ends in sand. The required length of sand coverage was obtained through a series of experiments. The effect of different configurations was studied by measuring the loss factor of the structure. The results obtained showed that the sand termination had a significant effect on the longitudinal modes of the pipe although this effect was independent of the coverage area. However, the circumferential modes were not effected at all. In addition, the effect of sand coverage over the ends of the pipe was investigated. It was shown that having the ends of the pipe open to the air or closed by sand had little effect on the modes of vibration of the structure.

1:45

**U4. Coal cutting noise control.** Mark R. Pettitt (Wyle Laboratories, P.O. Box 1008, 7800 Governors Drive West, Huntsville, AL 35807-5101) and William Aljoe (U.S. Bureau of Mines Pittsburgh Research Center, P.O. Box 18070, Pittsburgh, PA 15236)

The noise level near coal cutting operations in underground coal mines presently presents a significant health hazard to the miners. Hearing loss claims are a growing source of litigation in the industry. Through a series of research programs, the U.S. Bureau of Mines has developed methods to significantly reduce the noise generated during coal cutting. The noise control methods take full advantage of the dynamic characteristics of the forces generated during coal cutting. The spectrum of the cutting force was shown to be proportional to about one over the square of the frequency. Resonant response could, therefore, be significantly reduced by increasing the stiffness of the mining machine structures. Novel and effective techniques of achieving increased stiffness and damping without interfering with the machine's mining functions were developed and then demonstrated on selected types of coal mining machines. Finally, the feasibility of achieving cutting force isolation was established through laboratory cutting tool tests. In-mine viable isolated cutting tool designs were then developed and demonstrated through in-mine tests. As a result of this research work, the machine design concepts and techniques required to significantly reduce coal cutting noise and vibration, are now available to the coal mining industry.

2:00

**U5. The development of a computerized noise generator for the study of the effects of impulse noise on hearing.** R. Hétiu, C. Laroche (Ecole d'orthophonie et d'audiologie, Université de Montréal, CP 6128, Montréal, Quebec, Canada H3C 3J7), M. Sawan, and J. Nicolas (Département de Génie mécanique, Université de Sherbrooke, Sherbrooke, Quebec, Canada J1K 2R1)

A computerized noise generator is being developed in order to conduct parametric studies of the effects of impulsive noise on hearing. The parameters considered are the rise and the decay time, the amplitude, the repetition rate, and more especially, the frequency content. Basically, the spectrum of the desired signal is divided by the frequency response of an acoustic driver coupled to a power amplifier; the inverse Fourier transform of the result determines the corrected signal which allows us to generate an amplified replica of the desired signal. A variety of controlled transient signals can be, at this stage, reliably generated within the following limits: frequency bandwidth between 300 and 5000 Hz, decay times up

to 80 ms, peak amplitudes up to 140 dB SPL, and repetition rates between 0.1 to 10 p.p.s. Preliminary tests were conducted with a series of exposures to three impulsive noises of different frequency bandwidths: signal A extended from 300 to 1000 Hz, signal B from 300 to 3000 Hz, and signal C from 300 to 4000 Hz. Similar TTS growth curves were obtained when the peak SPLs of signals B and C were 10 to 12 dB lower than for signal A. The implications for damage risk criteria will be discussed considering that very slight variations in the time domain result in significant variations in the frequency content of impulse noise. [Work supported by I.R.S.S.T.]

2:15

**U6. Octave- and fractional-octave-band digital filtering based on the proposed ANSI standard.** Steven B. Davis (Signal Technology, Inc., 5951 Encina Road, Goleta, CA 93117)

A software implementation has been developed which is based on the proposed ANSI standard "Specification for Octave-Band and Fractional-Octave-Band Analog and Digital Filters." This standard is a revision of S1.11-1966(1976) "Octave, Half-Octave, and Third-Octave Band Filter Sets." Significant changes to the standard are the inclusion of digital filters, definition of octaves based on powers of 2 in addition to powers of  $10^{0.3}$ , and relaxation of filter symmetry requirements. The implementation is divided into three parts: (1) fractional-octave-band filter design, (2) an efficient decimation-in-time algorithm to perform the filtering operation on sampled signals, and (3) application of A-, B-, C-, and D-weighting curves. Numerical calculations and graphical displays of the output of the filters for white noise input verify the efficacy of the implementation. An evaluation of the digital filter design compared to the proposed stan-

dard, elucidates the difficulties in maintaining similar specifications in the analog and digital domains.

2:30

**U7. NASA aeronautical noise research.** Gerald G. Kayten, Stephen M. Wander (Aerodynamics Division, NASA Headquarters, Washington, DC 20546), and David Stephens (Langley Research Center, Hampton, VA 23665)

This paper presents a broad historical overview of NASA's major aircraft noise research program, and the advances and contributions made to the understanding, prediction, and reduction of aeronautical noise. Major programs including Quiet Nacelle, Quiet Engine, Refan, and the Aircraft Noise Prediction Program (ANOPP) are described and their role in the technology transfer process is discussed. It outlines projected future activities and discusses prospects for further noise reduction on conventional turbofan transports. New directions and focus in aeroacoustics research and technology for advanced turboprops, rotorcraft, and supersonic transports are also discussed.

2:45

**U8. Cockpit noise in light aircraft.** Fred C. De Metz, Sr. (1714 West 23rd Street, Suite A, Panama City, FL 32405)

Noise levels were measured in the cockpits of light civilian aircraft under various flight conditions and compared with OSHA noise standards. Dominant noise generation mechanisms are identified and the effectiveness of practical cockpit noise reduction techniques are explored.

WEDNESDAY AFTERNOON, 6 NOVEMBER 1985

DAVIDSON ROOM B, 1:00 TO 2:45 P.M.

## Session V. Physical Acoustics V: General Topics

F. Douglas Shields, Chairman

*Department of Physics and Astronomy, University of Mississippi, University, Mississippi 38677*

### Contributed Papers

1:00

**V1. An exact formulation for the internal pressure of a cavitation bubble.** Kerry W. Commander,<sup>a)</sup> Lawrence A. Crum, and Andrea Prosperetti (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677)

Recent experiments have shown that when the acoustic driving frequency is near one of the bubble's harmonic resonances, the theoretical values predicted by the Rayleigh-Plesset equation are inconsistent with observed values. This inconsistency lead Prosperetti to consider the internal pressure term in the Rayleigh-Plesset equation in a more general manner. In the past, the internal pressure of a bubble was assumed to be accurately predicted by a polytropic approximation. The internal pressure is computed from the conservation equations, resulting in a more accurate formulation. The new method also provides additional information about the internal thermodynamics of a bubble, which is explored in some detail. The two models are examined using some of the recent techniques in dynamical systems. "Feigenbaum trees" are shown for the two models of interest. This method for analyzing an equation is shown to be very sensitive to the internal pressure term, and thus it is an appropriate method for comparing different acoustic cavitation theories. [Work supported in part by NSF and ONR.] <sup>a)</sup> Present address: Naval Coastal Systems Center, Code 4120, Panama City, FL 32407-5000.

1:15

**V2. Prediction of tissue composition from ultrasonic measurements and mixture rules.** Robert E. Apfel (Department of Mechanical Engineering, Yale University, 2159 Yale Station, New Haven, CT 06520)

A methodology is presented for predicting the composition of tissues from measurements of the density, sound velocity, and acoustic nonlinear parameter, using mixture laws for the density, compressibility, and nonlinear parameter. It is shown that the mixture law for the nonlinear parameter plays an essential part in this methodology, which leads to the prediction of the volume fractions of water, protein, and fat in a given tissue. Data from the literature for solutions, blood, normal tissue, and cancerous tissue are investigated, and predicted fractions are consistent with tissue compositional information available in handbooks. More experimental work is needed with tissues of known composition in order to test more fully the proposed methodology. [Work supported by ONR.]

1:30

**V3. Angular equilibrium of an acoustically levitated sample.** M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)



In a triple axis levitator, three orthogonal fields position a sample at the center of a square cross-sectioned rectangular chamber. There are two torque mechanisms associated with this system. One is the well-known "rotating" torque whose magnitude depends on the phase angle between the two degenerate fields. A model for a second retarding torque was developed using Rayleigh disk theory applied to orthogonal plane-wave fields. The sample angular equilibrium position for zero-phase shift depends on the sample shape asymmetry and the relative magnitude of the applied pressures. For small phase angles, the sample adjusts to an angular equilibrium position where the "Rayleigh" retarding torque balances the "rotating" torque. For typical experimental conditions, the sample begins to rotate at a critical phase angle where the "rotating" torque overcomes the maximum "Rayleigh" torque. The dependence of the angular equilibrium position on the applied pressures, sample shape, and phase angle will be presented. A video tape of the phenomenon will be shown. [Work supported by NASA.]

1:45

**V4. The effect of finite phonon and roton mean free paths on the attenuation of fourth sound in HeII filled porous solid.** Steven R. Baker<sup>a1</sup> and Isadore Rudnick (Department of Physics, UCLA, Los Angeles, CA 90024)

Measurements of the attenuation of fourth sound in a HeII filled porous solid always show anomalously high values at low temperatures [M. A. Kriss, UCLA thesis (1969); H. J. Lauter and H. Wiechert, J. Low Temp. Phys. **36**, 139 (1979); S. R. Baker, UCLA thesis (1985)]. It is shown that this anomaly can be attributed to a reduction in the apparent normal fluid shear viscosity due to the finite mean free paths of phonons and rotons. Results of a model calculation of the mean free path correction to the attenuation of fourth sound in a HeII filled porous solid, which separately treats the average drift of phonons and rotons as the flow of an ideal gas through a porous solid [A. E. Scheidegger, *The Physics of Flow Through Porous Media*, Third Edition, 1974, p. 173], compare very well with measurements made in 1- $\mu$  packed powder down to 1.1 K at SVP, where the observed anomaly amounts to more than an order of magnitude. [Work supported by ONR.] <sup>a1</sup> Permanent address: Naval Postgraduate School, Code 61Xi, Monterey, CA 93943.

2:00

**V5. Particle behavior in a rotating ultrasonic waveguide.** Glenn Whitworth and Wesley L. Nyborg (Physics Department, University of Vermont, Burlington, VT 05405)

An apparatus was designed to allow a suspension of biological cells to be subjected to a well-defined, 160-kHz standing ultrasonic field while being viewed through a stereo microscope. The chamber, which has a square cross section and pressure release walls, acts as a single mode acoustic waveguide. Aqueous metrizamide solution is used to fill the ultrasonic chamber because it has a unique combination of properties including low viscosity, low osmolarity, and high density. The chamber rotates about its axis (whose inclination can be varied) producing the cen-

tripetal force necessary to contain the buoyant cells in the axial region. Observations were made on stroboscopically illuminated suspensions both of latex microspheres and of red blood cells. The particles arranged themselves at half-wavelength intervals into axially symmetric formations that became complex and flowering when many particles were present. Some aspects of this behavior are explained by preliminary theory that considers gravitational, rotational, and acoustic radiation forces on the particles. [Work supported by NIH grants GM0729409 (University of Virginia) and GM08209 (University of Vermont).]

2:15

**V6. Sonically produced heat in a fluid with bulk viscosity and shear viscosity.** Wesley L. Nyborg (Physics Department, University of Vermont, Burlington, VT 05405)

When a sound field is set up in an absorbing medium, heat is produced in a spatial pattern which, in general, depends on the absorption mechanism. Consider a fluid with bulk viscosity coefficient  $B'$  and shear viscosity coefficient  $\eta$ . In a continuous traveling plane wave the time-averaged volume rate of heat production  $\langle q_v \rangle$  at any point is proportional to the absorption coefficient  $\alpha$  which, in turn, is proportional to  $(B' + 4\eta/3)$ . In a more general field, the quantity  $\langle q_v \rangle$  can be calculated from a dissipation function if the components  $u_i$  of the velocity are known as functions of coordinates  $x_i$ . For a continuous field one obtains  $\langle q_v \rangle$  in the form  $(B'T_1 + \eta T_2)$ , where  $T_1$  and  $T_2$  are two different quadratic combinations of derivatives of the type  $\partial u_i / \partial x_j$ . Applying this expression to several examples (crossed plane waves, nearfield of a piston source) it is found that for a given value of  $\alpha$  the spatial distribution of  $\langle q_v \rangle$  is strongly dependent on the relative values of the coefficients  $B'$  and  $\eta$ . [Work supported by NIH Grant GM 08209.]

2:30

**V7. Sound amplification from controlled excitation reactions (SACER) in  $N_2/H_2$  mixtures.** F. Douglas Shields, Dadang Iskandar (Department of Physics and Astronomy, University of Mississippi, University, MS 38677), and Buford Anderson (Department of Physics, Murray State University, Murray, KY 42071)

Measurements of the resonant reverberation of sound in a closed tube have shown an increase in sound amplitude for 20 to 30 ms following the rapid excitation of the gas by an electrical discharge. The measurements have been corrected for sound absorption due to viscosity and thermal conductivity and the corrected amplification studied as a function of energy in the electrical discharge, vibrational frequency, gas pressure, and percentage of  $H_2$  in the mixture. The translational temperature during the relaxation process is monitored by measuring the sound speed in the tube. Amplification is attributed to the relaxation of the vibrational temperature which is left elevated to several thousand degrees following the electrical discharge. Experimentally measured gains have been compared to values previously predicted [F. D. Shields, J. Acoust. Soc. Am. **76**, 1749-54 (1984)]. [Work supported by the Office of Naval Research.]

**Session W. Psychological Acoustics II: Discrimination**

Larry Humes, Chairman

*Division of Hearing and Speech Science, Vanderbilt University School of Medicine, Nashville, Tennessee 37232***Contributed Papers****1:00**

**W1. Discrimination of spectral density.** W. M. Hartmann (Institut de Recherche et Coordination Acoustique Musicque, 31 rue Saint Merri, F75004, Paris, France, and Department of Physics, Michigan State University, E. Lansing, MI 48824) and Stephen McAdams (Institut de Recherche et Coordination Acoustique Musicque, 31 rue Saint Merri, F75004, Paris, France)

Experiments were performed to determine the ability of human listeners to discriminate between a sound with a large number of spectral components in a band, of given characteristic frequency and bandwidth, and a sound with a smaller number of components in that band. A pseudorandom placement of the components within the band ensured that no two sounds were identical. The data suggested that discrimination is primarily based upon the perception of temporal fluctuations in the amplitude envelope of the sound and secondarily upon a steady-state tone color. Experiments using clusters of complex sounds showed that listeners are able to use the information in harmonic bands to discriminate spectral density. [Work partially supported by NIH, the NSF, and the CNRS.]

**1:15**

**W2. The effects of changes in correlation on the discriminability of noise samples.** Susan M. Fallon and Donald E. Robinson (Department of Psychology, Indiana University, Bloomington, IN 47405)

Hanna [Percept. Psychophys. 36, 409–417 (1984)] has shown that wideband samples of reproducible noise are highly discriminable over a large range of durations. The discriminability can be reduced by increasing the correlation between pairs of samples to be discriminated within a same-different paradigm. In the present experiments, the second sample of the pair was generated by repeating a temporal segment of the sample presented in the first interval and appending to it a new segment of noise. The inter-pair correlation is then determined by the duration of the new segment. The total burst duration, as well as the duration and position of the appended segment (beginning, middle, or end) were manipulated. For all burst durations studied, position had a large and consistent effect: segments placed at the end were more discriminable than those in the middle which were more discriminable than those at the beginning. The importance of the temporal position of changes in the spectral-temporal profile of these broadband, noise waveforms will be compared to similar results obtained using other waveforms. [Work supported by AFOSR and NMRDC.]

**1:30**

**W3. Amplitude-modulated noise detection by goldfish revisited.** S. Coombs and R. R. Fay (Parmlly Hearing Institute, Loyola University of Chicago, Chicago, IL 02115)

Earlier studies on the ability of goldfish to detect amplitude-modulated (AM) noise signals [R. R. Fay, J. Neurophysiol. 44, 312–332 (1980)] suggested that goldfish were fundamentally different from other animals, including humans, in showing a flat modulation sensitivity function and no drop-off in sensitivity out to modulation rates of 400 Hz. To re-ex-

amine this anomaly, modulation sensitivity was measured in the goldfish at different sensation levels, using several experimental procedures and an improved technique for specifying the underwater acoustic stimulus at high modulation rates. Modulation sensitivity was found to be independent of sensation level at 10- and 100-Hz modulation. Sensitivity functions were relatively flat at modulation rates below 50–100 Hz with thresholds of AM detection occurring between 30% and 40% modulation. Greater than 60%–70% modulation was required for detection at 400 Hz. Randomly varying the overall intensity of unmodulated comparison noises and modulated noise signals produced little change in sensitivity or in the overall shape of the modulation function. These results show that the ability of goldfish to detect amplitude-modulated noise is not heavily dependent on intensity cues, nor fundamentally different from that of other species.

**1:45**

**W4. Capacity limitations for discrimination of isochronous and anisochronous tonal patterns.** C. S. Watson and D. C. Foyle (Hearing and Communication Laboratory, Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

In an earlier paper we reported that the discriminability of tonal patterns is strongly determined by the number of (sequential) components within the patterns. For patterns ranging in total duration from 62.5 ms to 2 s, accurate spectral-temporal resolution was possible only when patterns contained fewer than seven–nine tonal components [J. Acoust. Soc. Am. Suppl. 1 73, S44 (1983)]. Over this 32-fold range of total (and component) durations, patterns were nearly equal in discriminability when the frequency of a single component was varied by a fixed amount, or when randomly selected components were shifted from one sequential position to another. These results seem to demonstrate a limitation on the hearing of complex sounds that is based on total information content rather than on physical properties, specifically component duration. The previous experiments were conducted using an adaptive tracking procedure with  $n$  (the number of components in the patterns) as the dependent variable, and with the restriction that only isochronous patterns were presented. In two additional series of experiments it has now been shown that the same informational limits on pattern discrimination are obtained for anisochronous patterns and when the adaptive-tracking variable is the size of a frequency increment added to one of the pattern components. Possible meaning of this result for speech processing is considered. [Supported by NIH/NINCDS and by AFOSR.]

**2:00**

**W5. Factors affecting frequency discrimination in high- and low-pass noise backgrounds.** Deborah A. Fantini and David S. Emmerich (Department of Psychology, SUNY at Stony Brook, Stony Brook, NY 11794)

Previously, we presented data which indicated that frequency discrimination performance was poorer for tones presented near the spectral edge of a low-pass noise than for tones presented either near the edge of a high-pass noise or in an unfiltered white noise background. [D. S. Emmer-

ich, D. A. Fantini, and W. S. Brown, J. Acoust. Soc. Am. Suppl. 1 74, S34 (1983)]. The present study extends these findings in order to investigate how the steepness of the spectral edges of low- and high-pass maskers influences the discriminability of tones presented near these edges. Frequency separations corresponding to 75% correct responses were obtained in each of three low- and three high-pass noise backgrounds differing in the steepness of their filter skirts. The following results were obtained: (1) in the low-pass noise background, frequency discrimination performance improved as the filter skirt became more gradual, even though more noise power was added; (2) in the high-pass noise background, performance first improved and then became poorer as the filter skirt became shallower; (3) performance in low-pass noise was poorer than that in high-pass noise for the two steepest slopes employed (96 and 72 dB/oct) and better for the shallowest slope (36 dB/oct). Results are discussed in the context of a trade-off between possible edge effects and masking.

2:15

**W6. Basic auditory capabilities and resolving power for phonemes.** B. Espinoza-Varas and C. S. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Efforts to relate performance on speech tests to general auditory abilities have yielded conflicting evidence. One possible reason is that the resolving power for individual psychoacoustic dimensions is related more closely to the ability to resolve acoustic details of individual phonemes, than to *global* speech perception as assessed by the overall percent of tokens correctly identified on a speech test. Auditory capabilities were measured with a battery of psychoacoustic tests [Watson *et al.*, J. Acoust. Soc. Am. Suppl. 1 71, S73 (1982)]. Speech perception ability was studied with seven subtests of the CUNY Nonsense Syllable Test, NST (presented in quiet, 33 dB SPL). Indices of speech perception were derived for each individual phoneme, overall for each subtest, and overall for the seven NST subtests. Results from 34 normal listeners showed: (1) large differences in identification accuracy of phonemes within a subtest, which are not reflected in the overall measures; (2) strong biases towards specific phonemes within a subtest; and (3) differences in identification of an individual phoneme depending on the specific subtest in which it occurs. Thus the overall measures of speech processing are poor indicators of the ability to resolve specific phonemes. This result, together with the fact that phonetic distinctions are cued by multiple (redundant) cues, suggests that correlations between overall speech measures and auditory capabilities may be expected to be moderate or low. [Supported by NIH and AFOSR.]

2:30

**W7. Identification of voicing contrast via the tactile mode.** M. J. Collins, R. R. Hurtig, J. Besing, and D. J. Schum (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

It has been shown that tactile perception of speech can be categorical along a voice onset time (VOT) continuum and that the boundary between /ga/ and /ka/ occurs at a longer VOT value for tactile than for auditory perception [M. J. Collins and R. R. Hurtig, J. Acoust. Soc. Am. Suppl. 1 75, S33 (1984)]. In that study, a hearing aid with a bone conduction receiver had been used for tactile delivery of speech signals, and the question arose as to the uniqueness of the findings to the velar stop consonant and the specific device employed. The present investigation was designed to determine the consistency of the effect across place for stop consonants and across signal processing and transducer characteristics. An identification paradigm was used to determine the VOT boundary for three processor-transducer combinations along /ka-ga/, /ta-da/, and pa-ba/ continua. Boundaries fell at similar VOT values for each processor-transducer condition for velar and alveolar stimuli; complex interactions occurred for bilabials. It was concluded that both tactile system limitations and device characteristics were determinants of VOT boundaries. [Work supported in part by the Spelman Rockefeller Foundation.]

2:45

**W8. SDLB adaptation and an approximation of Steven's power law.** Ernest M. Weiler, David Sandman, Joseph Agnello, and Gene Balzer (Psycho-Acoustics Laboratory, Communication Disorders, Mail #379, University of Cincinnati, Cincinnati, OH 45221)

Initial curve fitting efforts having yielded similar results, data from four studies [E. Jerger, J. Acoust. Soc. Am. 29, 357-363 (1957); T. Palva and J. Karja, J. Acoust. Soc. Am. 45, 1018-1021 (1969); E. M. Weiler, M. Loeb, and E. A. Alluisi, J. Acoust. Soc. Am. 51, 638-643 (1972); and E. M. Weiler and J. D. Hood, Audiology 16, 499-506 (1977)] were combined to encompass intensities from 20 to 80 dB SL at 1 kHz. All studies used the classic method of simultaneous dichotic loudness balances (SDLB) as described by Hood [Acta. Oto-Laryngol. Suppl. 92 (1950)] and Small [Modern Developments in Audiology, edited by J. Jerger (Academic, New York, 1963)]. A regression line was fit to the log values of stimulus intensity and log adaptation (dB). Converting to the form of Steven's power law, the resulting constant  $K$  by this approach was 0.9449 and the exponent of intensity  $N$  was 0.73953. For loudness magnitudes,  $N$  has often been reported as about 0.60.

WEDNESDAY AFTERNOON, 6 NOVEMBER 1985

SUITE 4, 1:00 TO 2:45 P.M.

## Session X. Shock and Vibration II: Shock Response of Underwater Structures

Jerry H. Ginsberg, Chairman

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

Chairman's Introduction—1:00

### Invited Papers

1:05

**X1. Inelastic response of deeply submerged stiffened shells subjected to transient loads from explosions.** R. Daddazio, R. Atkash, and M. L. Baron (Weidinger Associates, 333 Seventh Avenue, New York, NY 10001)

The large deflection elastoplastic response of stiffened cylindrical shells subjected to transient loadings from explosions is considered. Because the cylinders are located at deep submergence, the effects of initial stress

distributions due to hydrostatic pressure are included in the analysis. A large deflection elastoplastic shell theory is utilized in the analysis. This shell theory defines a yield surface at each cross section of the shell in terms of the moment and direct force components at the section. An associated flow rule is utilized to define the plastic-strain rate. The EPSA (elastoplastic shell analysis) code is utilized in the analysis of the shell response to the dynamic loadings. Results obtained from the analysis have compared well with the results of experiments.

1:30

**X2. Finite element analysis of shock-induced hull cavitation.** Carlos A. Felippa and John A. DeRuntz, Jr. (Applied Mechanics Laboratory, Department 92-50, Bldg. 255, Lockheed Palo Alto Research Laboratory, 3251 Hanover Street, Palo Alto, CA 94304)

This paper describes the theoretical formulation and computational implementation of a method for treating hull cavitation in underwater-shock problems. In addition, the method can be applied to the analysis of submerged structures that contain internal fluid volumes. In the present implementation, the doubly asymptotic approximation (DAA) serves to simulate a radiation boundary that is located away from the fluid-structure surface at a distance sufficient to contain any cavitating region. The enclosed fluid is discretized with volume finite elements that are based upon a displacement-potential formulation. An explicit time-integration algorithm is used to advance the solution in the fluid-volume region, implicit algorithms are used for the structure and DAA boundary, and a staggered solution procedure has been developed to treat the interface condition. Results for two example problems obtained with the present implementation show close agreement with those obtained by other methods.

1:55

**X3. Numerical analyses of the linear interaction of pressure pulses with submerged structures.** H. Huang (Naval Surface Weapons Center, White Oak, Silver Springs, MD 20910)

This paper discusses the linear and nonlinear dynamic response of submerged structures to the impingement of pressure pulses, the strengths of which are such that their governing equations can be linearized. The range of validity of this linearization is assessed by comparing the linear and nonlinear solutions to a classical problem. The numerical analysis technique for cases where the structure is surrounded by a noncavitating fluid utilizes the boundary element representation of the surrounding wave fields in conjunction with finite element analysis of the structure response. The boundary element formulation is based on the exact Kirchhoff retarded potential integral solution to the linear wave equation. Therefrom a hierarchy of approximate boundary element formulations can also be obtained. The effectiveness of these formulations are examined by comparison of results to available classical solutions. The solution technique for cases of cavitating surrounding fluid requires the use of finite element representation of the exterior pressure field. Some effects of cavitation on the structural response are scanned.

2:20

**X4. Simple models for underwater shock based on modal analysis and DAA.** Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A general algorithm for evaluating the response of submerged structures to incident shock waves is developed based on a modal expansion version of the doubly asymptotic approximation (DAA) for fluid-structure interaction. A general procedure is outlined, in which the addition of substructures is treated in modular form. The structural motion is described in terms of the modes of *in-vacuo* free vibration of each substructure. Simple criteria for the companion fluid modes, which represent the effects of scattering and radiation, are developed. The nature of the modal coupling in this scheme, in combination with the fact that modal expansion techniques generally involve a comparatively small number of degrees of freedom, makes the approach very attractive for systems analysis. The procedure is illustrated by the case of a slender circular cylinder with spherical caps that is subjected to an exponential shock wave that arrives at an arbitrary angle of incidence. Such incidence, which was not considered in previous studies, is shown to lead to rigid body translation and rotation, as well as asymmetrical axial and bending deformations. [Work performed under the auspices of the U. S. Dept. of Energy by Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

## Session Y. Speech Communication IV: Speech Recognition

Cecil H. Coker, Chairman

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

## Contributed Papers

1:00

**Y1. Root-power sums and spectral slope distortion measures for all-pole models of speech.** Brian A. Hanson, Hynek Hermansky, and Hisashi Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

Distortion measures for use in speech processing are presented with emphasis on improvement of efficiency and recognition performance for noisy speech. Good results have been obtained from measures calculated from differences of root-power sums (i.e., differences of index-weighted cepstral coefficients:  $k \cdot c_k$ ). This type of distortion measure can be derived as an approximation to the spectral slope distortion measure, which expresses slope differences between test and reference logarithmic spectra. However, when used on all-pole model spectra, the root-power sum distortion is computationally more efficient. It is well-suited for use with both linear predictive (LP) analysis and the recently proposed perceptually based LP [H. Hermansky, B. A. Hanson, and H. Wakita, IEEE Proc. ICASSP 85, 509–512 (1985)]. The proposed distortion measures are shown to provide equal or better recognition scores compared to standard LP-cepstral matching in low noise conditions and substantial improvements in noisy conditions.

1:15

**Y2. A study of ambient and amplitude correction in isolated utterance recognition.** Subrata K. Das (IBM T. J. Watson Research Center, Computer Sciences Department, P.O. Box 218, Yorktown Heights, NY 10598)

Experiments demonstrating the importance of ambient and amplitude correction in an isolated word speech recognition system are reported. Whenever a speaker produces the speech input in a nonanechoic environment, such as in an ordinary office, the method of ambient correction helps to neutralize the effect of any change in ambient which would otherwise adversely influence the performance of the recognizer. On the other hand, the goal of amplitude correction is to compensate for amplitude level fluctuations encountered in any normal speech input. The experiments compare the performances with and without the corrections in the context of an experimental speech recognition system developed at IBM. A database consisting of the speech collected from three male and one female speaker is used for this study. Performance is measured by examining both the error rate and the time taken by the decoder. Significant improvement in performance is observed on both counts when the corrections are applied.

1:30

**Y3. Cues in the perception of infinitely clipped speech.** Daniel Kahn (Bell Communications Research, Morristown, NJ 07960)

"Infinitely clipped" ("IC") speech is surprisingly intelligible [Licklider and Pollack, J. Acoust. Soc. Am. 20, 42–51 (1948)]. Because infinite clipping so severely distorts the speech spectrum, it is sometimes speculated that a *time-domain* similarity between natural and IC speech—precise identity of zero-crossing locations—might account for the latter's intelligibility. This hypothesis has practical implications (should speech recognition algorithms look beyond power spectra?) and theoretical ones [cf. Scott, J. Acoust. Soc. Am. 60, 1354–1365 (1976)]. However, the hypothesis would be refuted if it turned out that the small amount of residual

similarity between the power spectra of the original and IC signals could alone account for the observed intelligibility. Three types of stimuli—natural vowels, /p,t,k/ in natural-sentence context, and synthetic vowels—were infinitely clipped ("IC stimuli"), then subjected to transformations which preserved the power spectrum but grossly distorted the phase spectrum, thereby destroying the original zero-crossing information. These transformed IC stimuli were no less intelligible than the unaltered IC stimuli, strongly suggesting that power spectrum rather than zero-crossing cues are responsible for IC-speech intelligibility. Further, constant-parameter, synthetic-vowel intelligibility decreased dramatically upon clipping, natural-vowel intelligibility relatively little, demonstrating the usefulness of spectral change and/or prosodic cues in the absence of accurate formant information.

1:45

**Y4. Adaptive length normalized DP matching method for connected word recognition.** Yasuhiro Matsuda and Toyohisa Kaneko (Science Institute, IBM Japan, Ltd. 5-19, Sanbancho, Chiyoda-ku, Tokyo, 102 Japan)

From a man-machine interface viewpoint, connected or continuous mode is much more preferred to isolated mode, but the former is much more difficult technically. A two-level DP [H. Sakoe, IEEE Trans. Acoust. Speech, Signal Process. ASSP-27, 588–595 (1979)] is a most widely used method for this problem. When we experimented with the two-level DP, we observed high error rates primarily due to the fact that the speaking speed often differs substantially between the training phase and recognition phase. To deal with this problem, our proposed method uses normalization of word length. Our method has two steps. The first step is to apply the conventional two-level DP. Then we normalize the length of each word and apply a two-level DP around the word boundaries obtained at the first step. We obtained the following improvement compared to a two-level DP with the following test data: 35 sequences of four-digit words by three male speakers with a single template for each digit. The average recognition rate in terms of digits, four-digit sequences, and the number of digits was improved by 1.9% (from 93.6% to 95.5%), by 6.7% (from 76.2% to 82.9%) and by 9.5% (from 88.6% to 98.1%), respectively. The computation time increased only about 10%.

2:00

**Y5. A practical implementation of speaker verification.** James W. Hicks, Jr. (SCI Technology, Inc., 8600 Memorial Parkway S.W., M/S-31, Huntsville, AL 35802)

This paper describes a pilot study in the implementation of a speaker verification system. Ten individual (seven males and three females) from the voice group at SCI served as enrollees for the verification system. In addition, four other group members acted as impostors/infiltrators in order to try to circumvent the system. The system employed was a Votan VPC-2000 in an IBM-PC. The enrollees to the system trained it with their name, employee number, and a password. The training was conducted on day 1 of the experiment and subsequently on days 14, 28, and 42. Each subject tested the system every working day for 2 months against each of their four training templates. Two of the four impostors were provided with the name, employee number, and password for each of the "authorized" subjects, while the other two impostors were not given this information. Each of the impostors attempted to access the system during the 2-

month experimental period. Data analysis is currently underway. Specific data and results will be discussed.

2:15

**Y6. The effects of instructions and feedback on speaker consistency in a speech recognition framework.** Linda A. Roberts, Jay G. Wilpon, Dennis E. Egan, and Jean Bakk (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Past research has demonstrated that a major problem in automatic speech recognition is speaker consistency. Instructions (suggestions of specific behaviors for consistent speech) and feedback (a visual display indicating the distance between each utterance and its template) were evaluated in a speaker-dependent recognition system as means of helping people speak more consistently. Utterances of 60 subjects were recorded using a  $2 \times 2$  design. Dependent measures included the difficulty of forming templates, distances between utterances and templates, and recognition accuracy. For every dependent measure, instructions had a positive effect on speaker consistency but feedback had no reliable effect. When subjects were retested 2 weeks after initial training, their speech was less similar to the templates than on the day the templates were formed, but the instructional effect was still reliable. An extensive analysis of speech errors was carried out. Judges subjectively classified each utterance according to different error categories. These subjective ratings were then compared to recognition accuracies. The error analyses suggested which

aspects of the instructions were effective and how future instructions might be improved.

2:30

**Y7. Application of an adaptive auditory model to speech recognition.** Jordan R. Cohen (IBM T. J. Watson Research Center, Continuous Speech Recognition Group, P.O. Box 218, Yorktown Heights, NY 10598)

One approach to designing signal processors for speech recognition has been to model the mammalian auditory system. Most designs have not attempted to capture the time-varying nature of the system, but have focused on the psychophysical aspects of critical bandwidth and loudness estimation. The IBM 5000-word speech recognition system [Bahl *et al.*, IEEE Trans. Pattern Anal. Machine Intell. PAMI-5, 179-190 (1983)] uses an auditory model in which psychophysical critical-band tuning and loudness estimation are combined with a firing-rate model patterned after that of Schroeder and Hall [J. Acoust. Soc. Am. 55, 1055-1060 (1974)]. The signal processing system consists of a critical-bandwidth filter bank, loudness estimation (intensity to the  $1/3$  power), and a reservoir-type firing-rate model with one internal state for each band. This model enhances transient events in the auditory signal, and causes rapid stimulus offsets to be marked by outputs smaller than the resting rate. The use of this auditory model in the IBM system produces a 4.4% error rate on a standard corpus of four speakers, while the previous filter-bank signal processor produces 7.4% errors on the same data.

WEDNESDAY AFTERNOON, 6 NOVEMBER 1985

SUITE 5, 2:00 P.M.

### **Meeting of Accredited Standards Committee S2: Mechanical Shock and Vibration**

to be held jointly with the

### **Technical Advisory Group (TAG) Meeting for ISO/TC 108 Mechanical Vibration and Shock**

P. H. Maedel, Jr. Chairman S2

*Westinghouse Electric Corporation, Lester Branch, P. O. Box 9175, Lester, Pennsylvania 19113*

G. Booth, Chairman, Technical Advisory Group for ISO/TC 108

*220 Clark Avenue, Branford, Connecticut 06405*

**Standards Committee S2 on Mechanical Shock and Vibration.** Working group chairpersons 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). There will be a report on the meeting of ISO/TC 108 and Subcommittees 1, 2, and 4, held from 2-13 September 1985, in Vienna, Austria.

**Session Z. Plenary Session**

Floyd Dunn, Chairman  
*President, Acoustical Society of America*

**Presentation of Tuning Fork Carried on Space Shuttle Mission 41-G, October 1984**

Paul D. Scully-Power  
*Naval Underwater Systems Center, New London, Connecticut 06320*

**Presentation of Awards**

**Presentation of the Pioneers of Underwater Acoustics Medal to Fred N. Spiess**

**Presentation of the Silver Medal in Physical Acoustics to David T. Blackstock**

**Presentation of the von Békésy Medal to Jozef J. Zwislocki**

## Session AA. Engineering Acoustics II: General

William Thompson, Jr., Chairman

*Department of Engineering Mechanics and Applied Research Laboratory, Pennsylvania State University, University Park, Pennsylvania 16802*

Chairman's Introduction—8:30

## Contributed Papers

8:35

**AA1. Computer-steered microphone arrays for sound transduction in large rooms.** J. L. Flanagan, J. D. Johnston, R. Zahn,<sup>a</sup> and G. W. Elko (Information Principles Research Laboratory, AT&T Bell Laboratories, Murray Hill, NJ 07974)

The quality of sound pickup in large rooms—such as auditoria, conference rooms, or classrooms—is impaired by reverberation and interfering noise sources. These degradations can be minimized by a transducer system that discriminates against sound arrivals from all directions except for that of the desired source. A two-dimensional array of microphones can be electronically beam steered to accomplish this directivity. This report gives the theory, design, and implementation of a microprocessor system for automatically steering a two-dimensional microphone array. The signal-seeking transducer system is implemented as a dual-beam, "track-while-scan" array. It utilizes signal properties to distinguish between desired speech sources and interfering noise. The complete automatic system has been tested in anechoic and medium-sized auditorium environments, and its performance is discussed. <sup>a</sup>Present address: Dornier System, D7990 Friedrichshafen, West Germany.

8:50

**AA2. A simple second-order toroid microphone.** J. E. West (AT&T Bell Laboratories, Murray Hill, NJ 07974) and G. M. Sessler (Technical University, Darmstadt, West Germany)

A second-order gradient microphone with a toroidal directional characteristic is described. The microphone consists of four commercially available, inexpensive first-order-gradient electret microphones which are arranged in the wall of a hollow cylinder at 90° angular spacings and whose outputs are added. The toroidal microphone shows a directional characteristic which is relatively frequency independent. It is characterized by rotational symmetry around the cylinder axis and by a cosine-squared dependence in the plane containing the rotational axis. In the direction of the axis, the sensitivity at midfrequencies is typically 20 dB lower than in the equatorial plane. The equalized frequency response in this plane is within  $\pm 3$  dB from 0.3–3 kHz, with a 1-kHz sensitivity of about  $-60$  dBV/Pa. The noise level of the microphone, measured in the frequency band 0.3–10 kHz, is  $-120$  dB re: 1 V, corresponding to an equivalent sound-pressure level of 34 dB.

9:05

**AA3. Ultrasonic depth gauge for liquids under high pressure.** Allan J. Zuckerwar (NASA Langley Research Center, M/S 238, Hampton, VA 23665), David S. Mazel, and Donald Y. Hodges (Old Dominion University, Norfolk, VA 23508)

An ultrasonic depth gauge of novel design continuously measures the level of a liquid subjected to a high pressure (up to 6000 psi), as is sometimes required for effective transfer of the liquid. The gauge operates as a composite resonator, fabricated from a standard high-pressure plug. A flat-bottomed hole is machined into the plug along its center line. An ultrasonic transducer is bonded rigidly to the interior surface of the bot-

tom wall, while the exterior surface is in contact with the liquid. Although the bottom wall is designed to satisfy the pressure code (ASME Code for Pressure Piping B31, 1980 edition), it is still sufficiently thin to permit ready excitation of the axisymmetric plate modes of vibration. The liquid depth is measured by a conventional pulse-echo technique. The external threads of the plug serve as the primary pressure seal. A prototype gauge, constructed with a 3/4-in. stainless steel plug and a 10-MHz piezoelectric transducer, was tested successfully in a 600-gallon water vessel at pressures up to 5200 psi.

9:20

**AA4. Experience with the two-microphone method for measuring the terminating impedance of a ducted burner during operation.** R. E. Howard (AT&T Information Systems, Holmdel, NJ 07733) and J. R. Mahan (Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

Previous papers on the two-microphone method establish its superiority over the classical impedance tube method for measuring the terminating impedance of a duct. The technique has been adapted here to measure the terminating impedance of a ducted burner with combustion noise acting as the sound source. The measurement system encounters a large dynamic range (30 dB) and high temperatures (900 K). Flow effects are negligible since Mach numbers remain below about 0.03. Results are compared to similar data from the literature, obtained using the classical impedance tube method, for duct terminating impedance in the presence of a hot flow. The method permits *in situ* measurement of the terminating impedance of a ducted burner during operation in spite of the hostile environment within the duct.

9:35

**AA5. Comparison of the thermoacoustic efficiency of premixed and diffusion hydrogen-flame ducted burners.** J. R. Mahan (Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061) and R. E. Howard (AT&T Information Systems, Holmdel, NJ 07733)

It has long been known that confining a flame within a duct has an influence on the efficiency with which it converts chemical energy to acoustic energy. The present paper describes an experimental program aimed at explaining and quantifying this influence. Results are presented for premixed and diffusion hydrogen-flame ducted burners of otherwise similar design operating in the 5- to 10-kW range. The results show that the premixed flame yields the higher thermoacoustic efficiency at a given thermal power level, and that its efficiency is sensitive to the power level. The thermoacoustic efficiency of the diffusion flame is lower and relatively insensitive to the power level. The results are explained in terms of Rayleigh's criterion and current combustion noise generation theories.

9:50

**AA6. The electroacoustic description of an underwater magnetohydrodynamic transducer.** Stephen C. Schreppler and Ilene



The Lorentz law, electromagnetic force in a conducting-dielectric, compressible, fluid medium, is investigated as a source of underwater acoustic radiation. The transducer system investigated consists of a duct which confines a conducting-dielectric fluid medium. A time-steady magnetic field and mutually orthogonal time-varying current density are established across the transverse dimensions of the duct. The system radiates from the longitudinal ends of the duct in a dipole, two-point array manner. The electroacoustic system equations in two port form, theoretical and experimental projector sensitivity, and pressure field directivity patterns will be discussed. [Work supported by ONR.]

10:05

**AA7. A coupled finite difference model for a fluid-loaded, free-flooded, thickness-polarized, piezoelectric cylindrical shell transducer.** Chwei-Chi Chiou and Peter H. Rogers (Department of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This paper presents a coupled finite difference model for a fluid-loaded, free-flooded, thickness-polarized, piezoelectric cylindrical transducer. In order to find the vibration response of this shell when subjected to a sinusoidal voltage drive, the Donnell's thin shell equations of axisymmetric free vibration of a circular cylinder are coupled with the piezoelectric canonical equations and a surface Helmholtz integral model for the acoustic radiation reaction. The acoustic loading and the piezoelectric effects are thus considered to be external loading factors in the vibration equations. Solutions for the velocities are then used as the inputs to a previously developed farfield radiation program (SHIP) to get the source levels over the frequency range, which depends on the specifications of the transducer under consideration. The validity of this model is verified by comparing the computed source levels with the experimental data for transducers of two different materials and four different geometrical dimensions.

10:20

**AA8. Derivation of the equations of motion of a piezoelectric accelerometer using the Lagrangian and modified piezoelectric stress equations.** Paul W. Jameson (Department 8C, BBN Laboratories, 10 Moulton Street, Cambridge, MA 02238)

The equations of motion of a piezoelectric accelerometer are determined by using the piezoelectric strain equations in inverse form to calculate the electromechanical stored energy in the system. In this form, the electric field and the stress are expressed as functions of the displacement vector  $D$  and the strain  $S$ . These latter quantities are used as the general coordinates for calculating the Lagrangian equation in a very straightforward manner. The mechanical and electrical impedance of the accelerometer is then computed and compared with experimental measurements of the sensor.

10:35

**AA9. Wave-vector-frequency filtering characteristics of multilayer plate systems via two-port methods.** Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881-0814)

The wave-vector-frequency ( $k - \omega$ ) filtering characteristics of multilayer systems provide a measure of the signal and noise transmission through such systems as sonar domes or flow noise decouplers. A two-port method of analyzing the  $k - \omega$  filtering characteristics of multilayer and elastic plate systems will be presented. Both liquid and elastic layers are represented using two-port parameters based on straightforward multidimensional Fourier transform methods. Timoshenko-Mindlin plate theory is utilized for the elastic layers. Multilayer configurations are readily addressed via the cascading of two-port matrices. Numerical results for the  $k - \omega$  filtering characteristics of several multilayer configurations will be presented. The physical phenomena which contribute to the structure of the transfer function in  $k - \omega$  space for the various configurations will be discussed.

**AA10. Nearfield calibration array utilizing piezoelectric polymer.** L. Dwight Luker, S. Petrie, and A. L. Van Buren (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337)

The Trott nearfield calibration array (NFCA) concept allows the far-field acoustic properties of underwater transducers to be determined in measurement facilities that are too small for direct farfield measurements. As a projector, the NFCA produces a nearly uniform plane wave throughout a large adjacent volume. The farfield sensitivity of a receiver can then be measured directly by placing it in the "plane-wave" volume. As a receiver, the NFCA measures only a single plane-wave component of the sound radiated by a projector located in the "plane-wave" volume. Existing NFCA's are primarily used as receivers and utilize small piezoceramic hydrophone elements arranged in planar or cylindrical configurations. The spacing between elements limits the upper frequency of operation of the NFCA; e.g., a 1-in. spacing allows operation up to about 50 kHz. This paper describes the design and construction of a planar NFCA utilizing a mosaic of nonelectroded sheets of the piezoelectric polymer PVDF attached to full area electrodes that subdivide the array into independently controlled annular elements. The PVDF NFCA was tested successfully as both a projector and a receiver by calibration of a high-frequency transducer. Although the available electronics limited the upper frequency of the test to 150 kHz, the PVDF NFCA should be useable, with suitable electronics, up to megahertz frequencies. [Work partially supported by ONR Code 220.]

11:05

**AA11. Sound intensity measurement in the air and on the surface.** O. H. Bjor (Norwegian Electronics a/s, N-1380 Heggedal, Norway) and R. J. Peppin (Scantek, Inc., 12140 Parklawn Drive, Suite 465, Rockville, MD 20852)

Sound intensity measurements require detection of both sound-pressure and sound particle velocity. The sound pressure is easily measured with a condenser microphone. The sound particle velocity can be detected indirectly by the conventional pressure gradient method using two closely spaced microphones, the spacing of which is a function of frequency. Alternatively, particle velocity can be detected directly by a new microphone that uses the interaction of the audible sound field and an ultrasonic wave. This new principle allows the construction of a sound intensity probe with a very broad frequency range (20 Hz–5 kHz) without readjustments to the probe. The particle velocity detection principle may also be used for noncontact measurement of surface velocity. As a result, surfaces of any materials may be explored with the probe without any direct contact, eliminating the limitations (such as surface loading, poor mounting methods, and sensitivity problems) posed by conventional accelerometry techniques.

11:20

**AA12. Numerical study of an area array for nearfield sound-pressure measurement.** N. Yen and Robert D. Corsaro (Physical Acoustics Branch, Naval Research Laboratory, Washington, DC 20375-5000)

In a previous work [M. P. Hagelberg and R. D. Corsaro, J. Acoust. Soc. Am. 77, 1222–1228 (1985)], we described our experimental experience using a planar polymer hydrophone for area-averaged acoustic pressure measurements in the nearfield of an acoustic source. This present paper addresses theoretically the accuracy of such a measurement. A numerical analysis based on the Rayleigh integral formulation is performed on the nearfield radiation from a circular projector (disk) with  $ka$  from 10–40. The spatial average of the acoustic pressure received by an area array in the front of this projector is computed in terms of the covered area parallel to the projector's surface. Its magnitude is shown to vary only slightly with changes in the array's location, even within the projector's nearfield region. The reflected wave from a plate of finite thickness is also investigated within this nearfield region by means of the image method. These simulated results are compared with experimental echo reduction measurements obtained under plane-wave conditions.

**AA13. Reciprocity calibration in a compliant cylindrical tube.** Michael B. Johnson and Steven L. Garrett (Department of Physics Naval Postgraduate School, Monterey, CA 93943)

A new method for absolute calibration of electroacoustic transducers is described which permits low-frequency reciprocity calibration in a test apparatus of small dimensions. The method takes advantage of the reduced wave speed in a cylindrical water column bounded by a compliant PVC tube to reduce the length of the standing wave to dimensions which can be easily handled in the laboratory. Results of a limited series of experiments show a mean reproducibility of  $\pm 0.8$  dB in a 1-m-long tube at frequencies between 750 and 1100 Hz and sound speeds from 330–365 m/s. The hydrophone voltage sensitivities obtained by the method, however, systematically differ by 1.5 dB from those measured by comparison with a standard hydrophone. Additional research will be required to resolve this discrepancy. [Work supported by ONR.]

**AA14. Improvement to the transfer function method for determining the complex dynamic modulus of polymer composites.** Arthur E. Sempale and

William Thompson, Jr. (Department of Engineering Science and Mechanics and The Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

Measurements of loss factor and storage modulus of diphasic transducer composites of polymers and piezoceramic were made using the transfer function method. This method consists of exciting a mass-loaded rod into longitudinal vibrations. The complex acceleration ratio between its ends is related to the complex modulus by two coupled, transcendental equations derived from the solution of the longitudinal wave equation with appropriate boundary conditions. In theory, these equations can be solved at any frequency by an iterative procedure. In practice, convergence problems reduce the valid solutions to those just at the longitudinal resonance frequencies. Improvements made during this study enable the equations to be solved at any frequency, except where the phase of the acceleration ratio is zero. Plots of loss factor and storage modulus are presented for a number of these new composites and other high-damping polymeric materials. A comparison between present results and those quoted in other studies indicates that this method produces reliable results. The improvements enable results to be obtained more quickly and with much less effort than previously required. [Work supported by ONR.]

THURSDAY MORNING, 7 NOVEMBER 1985

REGENCY BALLROOM IV, 8:30 TO 11:45 A.M.

## Session BB. Speech Communication V: Speech Production II

James E. Flege, Chairman

*Department of Biocommunication, University of Alabama Medical Center, University Station, Birmingham, Alabama 35294*

### Contributed Papers

**BB1. The effects of phonation on  $^{133}\text{Xenon}$ -inhalation air curves (of the kind used in deriving regional cerebral blood flow).** C. Formby<sup>1</sup> (Departments of Communicative Disorders and Neurology, University of Florida, Gainesville, FL 32610), R. G. Thomas (Department of Statistics, University of Florida, Gainesville, FL 32610), W. S. Brown (Institute for the Advanced Study of Communication Processes, University of Florida, Gainesville, FL 32611), and J. H. Halsey, Jr. (Department of Neurology, University of Alabama—Birmingham, Birmingham, AL 35294)

Regional cerebral blood flow (rCBF) may be measured by noninvasive inhalation techniques that use end-tidal values of respired air radioactivity to estimate the isotope concentration in arterial blood. These end-tidal data are used as an input function in a mathematical equation for measurement of rCBF. For a subject with normal pulmonary function, end-tidal air is assumed to be in equilibrium with the arterial blood at the alveolar surface of the lung during passive breathing. However, an assumption of equilibrium may not be valid for active breathing conditions such as those during phonation. In light of the important role of the respired air curve in the measurement of rCBF by inhalation methods and the possible confounding effects of phonation, respired air curves have been analyzed and will be presented for three groups of male college students: (1) vocalists trained by the same voice instructor ( $n = 12$ ); (2) members of the same University choir ( $n = 13$ ); and (3) nonmusicians ( $n = 12$ ), for three conditions of phonation (of the national anthem). These conditions are (1) speaking, (2) singing, and (3) humming. The curves have also been analyzed for passive breathing. [Portions of this work were supported by an NIH Grant to UAB.] <sup>1</sup> Data were collected while C. Formby was a postdoctoral fellow in the Department of Neurology at the University of Alabama—Birmingham.

**BB2. Rule-controlled data base search.** Rolf Carlson and Björn Granström (Department of Speech Communication and Music Acoustics, Royal Institute of Technology, S-100 44 Stockholm, Sweden)

In current acoustic-phonetic research, there is a need for huge speech data bases. There are considerable problems in administering such data bases, both to transcribe and segment the speech and to easily access stored material. We have created a speech analysis system to attempt to alleviate these problems. Speech data are stored in sentence-sized files. These files are segmented and transcribed semiautomatically [M. Blomberg and K. Elenius, *STL/QPSR* 1/(1985)] given a phonetic transcription of the utterance. This transcription is generated by the text-to-phonetic component of our synthesis by rule system [R. Carlson and B. Granström, *IEEE-ICASSP* (1976)]. The same rule structure, similar to the notation used in generative phonology, is used for accessing the data. By a brief rule statement, speech segments meeting the specified contextual conditions can be identified. Durational data can be collected directly during the data base search. Spectral analysis programs, operating with a variety of spectral representations [R. Carlson and B. Granström, Eds., *The Representation of Speech in the Peripheral Auditory System* (Elsevier/North Holland)] have also been created that display the result, typically as a mean/s.d. spectrum or contour histogram spectrum.

**BB3. Generation of vocal-tract shapes from formant frequencies.** Donald C. Wold (Department of Physics and Astronomy, University of Arkansas at Little Rock, 33rd and University Avenue, Little Rock, AK 72204)

A microcomputer program has been developed to generate midsagittal vocal-tract shapes from the corresponding formant frequencies. The concept of formants is usually presented in textbooks and references for courses in the science of sound. The frequencies of these resonances are used to obtain the vocal-tract configurations for different vowel positions of the tongue. Ladefoged *et al.* [J. Acoust. Soc. Am. 64, 1027-1035 (1978)] have devised an algorithm that uses only the first three formant frequencies of vowels for generating vocal-tract shapes as seen on midsagittal x-ray diagrams. Ladefoged *et al.* gave a procedure for estimating the width of the vocal tract at different positions, but not the two-dimensional shape. In order to graph the midsagittal vocal tract shapes with a microcomputer having limited memory, an arbitrary, standardized, upper surface of the vocal tract was constructed. The equation of a catenary was used as a reference curve. A parametric form was found useful for generating graphs and relating path length to distance along the vocal tract.

9:15

**BB4. Frequency of acoustical patterns in Spanish syllables.** Miguelina Guirao and María García Jurado (Laboratorio de Investigaciones Sensoriales, CONICET, CC 53, 1453 Buenos Aires, Argentina)

This work aims at a better understanding of the internal acoustical structure of Spanish speech sounds. A sample of 43 000 words was used to analyze the frequency of occurrence of the main spectral patterns of 68 141 syllable-sized segments. Acoustical syllabic components (ASC) were classed in periodic nonvocalic (P) and vocalic (V) sounds, bursts (B), and noise sounds (N). The first computer-aided count gave the frequency of syllabic types in conjunction with their acoustical classes. Within type CV (55.8%) combinations of ASC were B + V 26.8%, P + V 18.6%, and N + V 10.4%. Within type CVC (15.8%) predominant combinations were B + V + P 5.5%, P + V + N, and B + V + N each one 2.7%. A second count gave the incidence of ASC in the first 3996 words. Combinations were P + V 29% and B + V 24%. Vowels sounds were 14%. The most frequent ASC in initial and final word position were registered. Starting words B + V are 40% and vowels 25%; terminating words B + V are 45% and P + V 30%. Relative occurrences of B + V in both positions indicate their high functional load at setting acoustical boundaries between Spanish words.

9:30

**BB5. An acoustic and perceptual study of the voiceless portion of unvoiced stop consonants in American English.** Carol Chapin Ringo (Research Laboratory of Electronics, Room 36-511, Massachusetts Institute of Technology, Cambridge, MA 02139)

An investigation of the acoustic properties of the voiceless portion of unvoiced stops was undertaken. Three male native speakers of American English were recorded reading a list of "CVd" words, in which the consonants /p,t,k/ appeared before the vowels /i,j,e,E,ae,u,u,Λ,o,a/. For each speech token, formant frequencies and amplitudes at 10-ms intervals were estimated visually from hard copies of computer-implemented DFT displays. It was found that for certain CV combinations the ratio of the average amplitude of F3 to that of F2 during the aspiration interval is considerably greater than the same ratio computed during the vowel. This finding could be explained either by greater losses through the open glottis for the second formant, or by the different excitation of the formants as a consequence of the source location and spectrum. In some cases, the presence of substantial energy in the region of F4 and F5 favors the latter explanation. These results raise questions concerning the perceptual importance of F2 in identifying place of articulation in voiceless stop consonants. Preliminary perceptual experiments are described in which this issue is explored. [Work supported by a grant from NINCDS.]

9:45

**BB6. Spectrographic analysis of vowels in Down's syndrome speech.** Mikael D. Z. Kimelman, Edie Swift, Margaret M. Rosin, and Diane M. Bless (Waisman Center on Mental Retardation and Human

Development and Department of Communicative Disorders, University of Wisconsin—Madison, Madison, WI 53705)

The speech of individuals with Down's syndrome is often considered to be worse than that of individuals with other forms of developmental disabilities. The present study was initiated to shed light on a broad range of communication abilities of individuals with Down's syndrome. Preliminary evidence in the form of spectrographic analysis of vowels will be presented. Several factors including formant patterns and variability will be reviewed. These data will be discussed with respect to possible explanations for this population's poor speech intelligibility. [Work supported by Award from the National Institute of Education No. NIE-G-81-0009.]

10:00

**BB7. More on the speech production deficit associated with Parkinson's disease.** Gary Weismer, Mikael D. Z. Kimelman, and Steven Gorman (Speech Motor Control Laboratories and Department of Communicative Disorders, Waisman Center, University of Wisconsin—Madison, Madison, WI 53705)

For certain patients with Parkinson's disease, we have shown that aspects of vowel production inferred from acoustic analysis are consistent with the akinetic and bradykinetic characteristics of Parkinsonian limb movements [G. Weismer and M.D.Z. Kimelman, J. Acoust. Soc. Am. Suppl. 1 77, S87 (1985)]. Because Parkinsonian dysarthria is often characterized by abnormally fast articulation rates, it is possible that some of the reduced transition extents and rates observed for Parkinson's patients do not reflect directly the nature of the speech movement disorder, but rather are by-products of the abnormally fast rates. To address this issue, vowel transition data will be presented from three groups of speakers (Parkinson's patients, young adults, and geriatrics) who tend to separate on the variable of articulation rate. In addition, the effect of fast rate productions on transition extents and rates in normal speakers will be discussed as a potential simulation of Parkinsonian dysarthria. [Work supported by NIH Award No. NS 13274.]

10:15

**BB8. Lip/jaw coordination with larynx in the speech of stutterers.** Gloria J. Borden and Joy Armson (Temple University, Philadelphia, PA 19122 and Haskins Laboratories, New Haven, CT 06510)

Laryngeal movements were inferred from slow and fast (filtered) impedance changes recorded by an electroglottograph and lip/jaw movements from the deflections of an LED attached to the lower lip. Stutterers and control subjects repeated two, four-digit numbers five times each or until they were judged to be fluent. The interactions of the lip/jaw lowering at the release of the [f] constriction in the words "four" and "five" with vocal fold adduction for the vowel and the onset of voicing were analyzed by making selected temporal measures in both the displacement and the velocity traces. Results of the data analyzed thus far indicate functional relationships: i.e., the lip/jaw system coordinates with vocal folds for the [f] by closely timing oral constriction release with maximum vocal fold opening, and for the vowel by closely timing peak velocity of lip/jaw opening with peak velocity of vocal fold adduction. Stutterers show aberrant laryngeal-supralaryngeal interactions as well as instances of "normal" coordination in the midst of stuttering. [Work supported by NIH.]

10:30

**BB9. Deaf speaker's control of acoustic intensity.** James Mahshie, Fred Brandt, and Barbara Brunner (Department of Audiology, Gallaudet College, Kendall Green, Washington, DC 20002)

Control of acoustic intensity is required for both linguistic and paralinguistic aspects of speech communication. While generally recognized that deaf speakers experience difficulty learning to control acoustic intensity, the characteristics and mechanisms of intensity control among deaf speakers are not well understood. The objectives of the present research were to: (1) study and describe characteristics of intensity control by deaf individuals during speech production; (2) compare deaf and normal-hear-

ing speaker's control of acoustic intensity; and (3) suggest possible mechanisms used by deaf speakers to control acoustic intensity. Deaf and normal-hearing speakers produced sustained vowels at different acoustic intensity and fundamental frequency levels. The average airflow, acoustic intensity, spectral characteristics, fundamental frequency, and characteristics of vocal fold oscillation were measured. Results showed that deaf speakers tended to use greater than normal air flow for all speaking tasks. While normal-hearing speakers tended to increase airflow with increased intensity, the deaf speakers often failed to produce such a distinction. However, certain systematic changes in laryngeal adjustments associated with increased acoustic intensity were observed in both groups of speakers. Possible mechanisms for intensity control will be discussed, along with clinical implications.

10:45

**BB10. Vision in lip positioning skills of children with normal and impaired hearing.** S. G. Fletcher (Biocommunication Department, University of Alabama in Birmingham, Box 187 University Station, Birmingham, AL 35294)

Interplay between visual feedback and lip and jaw positioning skill was studied in twenty 5- to 14-year-old children with normal hearing, and with severe-to-profound hearing impairment. With visual feedback, the subjects in both groups were similar in response time and in accuracy of matching six visually specified lip separation "targets." Special skill in processing visual information by the hearing-impaired subjects was suggested by higher velocities in lip movements toward the targets and shorter latencies in reaching the goal positions. In the responses of the hearing children, lip-closing movements were executed more accurately than opening movements, both with and without visual feedback. This was found only in the nonfeedback condition with the HI children, suggesting that the visual information had a greater effect on their responses. In general, the findings showed that, given visually displayed lip position targets and feedback from positioning actions, children can achieve them with high accuracy, regardless of hearing status or amount of prior speaking experience.

11:00

**BB11. Long-term average spectra of continuous speech before and after Tomatis audio-vocal training.** William Weiss (Department of Theatre, University of Ottawa, Ottawa, Ontario K1N 6N5, Canada)

Three subjects have undergone the Tomatis audio-vocal training, which is a method that filters sidetone for auditory feedback. Long-term average spectra of voiced and whispered speech was taken before and after the training with the following results: (1) Decrease in energy in the 50- to 200-Hz zone; (2) increase in energy in the 800- to 1200-Hz zone; (3) different individual overall energy shifts; (4) larger and wider formantic peaks in

the long-term average spectra; (5) increase in global energy; (6) greater pitch and power variability; and (7) increase in average pitch. A discussion concerning the employment of auditory feedback techniques in voice training follows. Future research should determine which psychophysiological characteristics of individuals determine their reactions to auditory feedback. It is also necessary to establish acoustic standards that could indicate desirable directions for voice training.

11:15

**BB12. Vowel  $F_1$  as a function of speaker fundamental frequency.** Ann K. Syrdal (University of Texas at Dallas, Callier Center, 1966 Inwood Road, Dallas, TX 75235) and Shirley A. Steele (University of Texas at Dallas, and Texas Instruments, Inc., P. O. Box 226015, M/S 238, Dallas, TX 75266)

The relationship between speaker fundamental frequency ( $F_0$ ) and first formant frequency ( $F_1$ ) was investigated. A wide  $F_0$  range was sampled in each of two sentence positions, using representative high and low vowels (/i/ and /ae/) in two phonetically controlled, interchangeable key words. Data were collected from four women and four men and processed with a 14th-order LPC analysis, using a correlation pitch tracker. The bark-transformed  $F_0$  ranges were equivalent for women and men. The relationships between  $F_0$  and  $F_1$  were similar for both sexes, although women consistently adjusted  $F_1$  frequency in /i/ to closely approximate  $F_0$  in their higher  $F_0$  range, whereas men typically did not. It was found that  $F_1$  increased significantly with increasing  $F_0$ . Although the difference between bark-transformed  $F_1$  and  $F_0$  decreased significantly as  $F_0$  increased, a 3-bark critical distance consistently separated /i/ and /ae/ throughout the  $F_0$  range. [Work supported in part by NIH.]

11:30

**BB13. Frequency and amplitude perturbations of the human voice determined with a reflectionless tube.** Claire W. Anderson, A. Yonovitz (Speech and Hearing Institute, The University of Texas Health Science center, Houston, TX 77030), and Wallace L. Anderson (Department of Electrical Engineering, University of Houston, Houston, TX 77004)

In this study, the glottal function was obtained through the use of a reflectionless tube. This tube acted as a pseudoinfinite termination of the vocal tract. Male subjects phonated a neutral vowel (schwa) while target matching to tones throughout the vocal range. These waveforms were digitized at 61 440 samples per second for 1 s. Amplitude and frequency perturbations were obtained for cycle-to-cycle variations, as well as a normalized distribution. Spectral analysis of the digitized waveforms provided a simultaneous description of frequency modulation (jitter) and amplitude modulation (shimmer). Frequency perturbations monotonically increased (approximately 1%) with target frequency. Amplitude deviations will be discussed as they relate to the frequency modulation.

## Session CC. Underwater Acoustics IV: Scattering

Finn B. Jensen, Chairman

SACLANT ASW Research Centre, I-19026 La Spezia, Italy

Chairman's Introduction—8:30

## Contributed Papers

8:35

**CC1. High-frequency underice acoustic backscattering: A simulation model study.** Garner C. Bishop and Leonard E. Mellberg (Naval Underwater Systems Center, Newport, RI 02841-5047)

A simulation model that evaluates the backscatter of a high-frequency acoustic pulse from the underice surface of pack ice regions, characteristic of the interior Arctic, has been used to investigate the extent to which various morphologic, geometric, and acoustic parameters, used to describe the underice surface, influence the backscatter. The simulation model utilizes a three-dimensional underice canopy synthesized from two-dimensional acoustic profile data. The large scale scattering features of the underice surface are modeled as first-year ice keels and sloping flat ice regions. The small scale surface structure is modeled as an ensemble of randomly oriented ice blocks. The Kirchhoff approximation is used to evaluate the scattering strength of an individual ice block which, in turn, is used to calculate the scattering strength of the underice surface. The sonar equation is used to calculate reverberation. The parameters used in the model may be divided into four groups: canopy, ice keel, ice block, and acoustic parameters. Parameters used to identify and classify the constituent ice features of the canopy are varied, and it is shown that, for physically reasonable parameters, reverberation is essentially independent of the feature identification and classification criteria. Reverberation is also shown to be largely independent of the parameters that describe only the large scale structure of the ice keel, but is strongly dependent upon parameters that effect the area and spatial orientation of individual ice blocks.

8:50

**CC2. The influence of water-to-ice transition layers on the reflection coefficient and high-frequency acoustic backscatter from an ice keel.** Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529)

The physical properties of ice have been included in a plane-wave reflection coefficient calculation that is then used in a model to predict high-frequency acoustic backscatter from an ice keel. Reflection coefficients were computed using a plane-wave model which had parallel layers of varying thicknesses. Each layer was homogeneous in density, compressional velocity and attenuation, and shear velocity and attenuation. Roughness could be included on any layer. A profile of the ice was used [Branch and Posey, *J. Acoust. Soc. Am. Suppl. 1* 77, S56 (1985)] that allowed a gradual transition from water to multilayered ice. The ice was backed by water to simulate conditions that exist on ice blocks, which form ice keels. An ice-keel model [Bishop *et al.*, *J. Acoust. Soc. Am. Suppl. 1* 77, S56 (1985)] was modified to accept oblique incidence signals and different types of ice-block rotations. The reflection coefficients were used in the ice-keel model to calculate target strength. Examples are presented that indicate that the reflection coefficient can, at low grazing angles, be highly dependent upon the water-to-ice transition layers. This can give higher returns from the ice-keel model than expected. Moreover, the reflection coefficient's influence on the ice-keel model is dependent upon the type of ice-block rotation used. [Work supported by NAVSEA.]

9:05

**CC3. Comparison of ultrasonic measurements of acoustic scattering from hard infinitely thick striated surfaces with Twersky theory.**

T. McClanahan (Naval Air Systems Command, Washington, DC 20361), O. Diachok, and S. Wales (Naval Research Laboratory, Code 5120, Washington, DC 20375-5000)

Ultrasonic laboratory scale measurements of scattering from an infinitely thick, striated surface were conducted to test the accuracy of the Twersky theory predictions. These measurements represent the first phase of a measurement program that is intended to provide a baseline and stimulus for refined theories of underice reflectivity and scattering, and will, in its next phase, consider randomly distributed and oriented half-cylinders on plates. Results of the initial phase reveal that measured reflection coefficients are in excellent agreement with theory at grazing angles between  $10^\circ$  and  $50^\circ$ , when  $ka < 1$  and when  $ka \gg 1$ , where  $k$  is the wavenumber and  $a$  is the average half-cylinder depth. At intermediate frequencies, measured losses were substantially greater than predictions due to resonance scattering associated with the physical/geometrical properties of the half-cylinders. These results imply that both theoretical and laboratory scale modeling of scattering from sea ice at intermediate frequencies must consider both geometrical and physical properties of the proturbances. [Work supported by ONR.]

9:20

**CC4. A multibounce, single-scatter ray theoretic model for underice predictions.** A. Tolstoy, D. H. Berman, E. B. Wright, and R. N. Baer (Naval Research Laboratory, Code 5120, Washington, DC 20375-5000)

This study describes the propagated and scattered acoustic field from a point source to a point receiver in an approximate underice environment characteristic of the deep Arctic Ocean. Our model considers both the azimuthal and vertical redistribution of scattered energy caused by a rough surface overlying an upwardly refracting, linear sound-speed profile. The model considers one nonspecular surface scattering per path (connecting the source with the receiver) coupled with any number of specular interactions, each of which suffers losses due to scattering into nonspecular directions, i.e., consistent with conservation of energy. The propagation model is based on ray theory, while the scattering model uses Twersky's scattering cross section for hard or soft planar surfaces roughened by hard or soft, hemispherical bosses. The model calculates intensity as a function of position and direction. Bistatic results at all frequencies show that for hard and soft surfaces the strongest scattered field is produced directly above the submerged source, and that the total scattered energy levels increase as frequency increases. Monostatic geometries and effects of distributions of boss sizes will also be considered. [Work supported by NRL and ONR.]

9:35

**CC5. Source localization in range and depth in an Arctic environment.** R. G. Fizell and S. C. Wales (Code 5120, Naval Research Laboratory, Washington, DC 20375)

A fixed cw source in the Central Arctic was accurately localized in range and depth using wave-based full-field ambiguity function techniques. The data were taken during the Fram IV experiment, utilizing a vertical array spanning the upper 960 m of a deep-water environment (generally greater than 2800 m) and a high SNR signal from a 20-Hz source at a distance of 270 km from the array, both suspended from the ice cover. The predicted field was computed using a fast field program. Both a

linear correlation of the predicted field with the received field and a maximum likelihood method (MLM) estimator correctly localized the source. The MLM estimator produced sidelobes significantly below the main peak, while the linear correlator produced sidelobes sufficiently large to be taken as false targets. The linear processor, however, was less sensitive to mismatch between the predicted and measured field. Array gain for the linear estimator was found to be 2 dB higher than for a conventional plane-wave beamformer.

9:50

**CC6. Elastic wave scattering in a horizontally stratified ocean environment with application to arctic propagation.** W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375) and Henrik Schmidt (SACLANT ASW Research Centre, I-19026 La Spezia, Italy)

A previously developed boundary perturbation method [W. A. Kuperman, J. Acoust. Soc. Am. 58, 365 (1975)] is extended to treat scattering at a randomly rough interface which separates viscoelastic media. This method is then combined with a full-wave treatment of sound propagating in a stratified ocean described by a system of liquid and elastic layers [H. Schmidt and F. B. Jensen, J. Acoust. Soc. Am. 77, 813-825 (1985)]. The net result of combining the extended boundary perturbation method with the full-wave solution technique is to define a set of effective potentials which, when inserted in the full-wave solution algorithm, yields the coherent components of the compressional and shear wave fields in the Kirchhoff approximation. The fields decay with range due to boundary roughness scattering into both incoherent compressional and shear waves. A natural outcome of this model is also the option to calculate the coherent reflection coefficient for an arbitrarily arranged layered system of liquid/solid media separated by randomly rough interfaces. Numerical examples of reflection coefficients and sound propagation in an ocean described by a stratified waveguide are presented. Emphasis is on arctic propagation demonstrating a new low-frequency attenuation mechanism resulting from scattering into shear waves at the upper and lower rough boundaries of an ice layer.

10:05

**CC7. Resonant scatterer configurations near elastic boundaries.** I. Tolstoy (Knockvennie, Castle Douglas, SW Scotland)

Systems of precisely spaced bubbles, air-filled shells or balloons in water, insonified at frequencies near their intrinsic radial resonance  $\omega_0$  exhibit *true resonant modes*. For resonant wavenumber  $k_R$  (in water) and scatterer radius  $a$ , this leads to amplification factors  $(k_R a)^{-1}$  above and beyond the similar classical factor due to single scatterer, and thus to net pressure amplification of order  $(k_R a)^{-2}$  relative to the incident field (i.e., over 120 dB on the intensity scale, on or inside the scatterer). This effect is predicted by formulas for the equivalent source strength  $B$  of each scatterer of the system, account being taken of multiple scatter interaction. Whereas for pairs of scatterers, or periodic lattices,  $B$  is known to exhibit relatively modest maxima for selected values of  $x = kl$  ( $l$  being the spacing between scatterers), which had been called resonances [V. Twersky, J. Opt. Soc. Am. 52, 145-171 (1962)], one can show that, under certain conditions, *true resonances* exist—i.e., in the absence of attenuation,  $B$  exhibits *real poles*. This phenomenon is much enhanced for systems near elastic boundaries for which coupling between scatterers is mediated by surface waves, and doublet or triplet configurations develop spectra of resonant configurations  $x_n$  at frequencies  $\omega_n$  (the latter being all close to  $\omega_0$  for a spectrum of scatterer sizes  $a_n$ ). [Work supported by ONR.]

10:20

**CC8. Rough surface scattering: Exact numerical methods compared with the Kirchhoff approximation.** Eric I. Thorsos (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, 1013 N.E. 40th Street, Seattle, WA 98105)

Acoustic scattering has been calculated exactly using integral equation methods for deterministic and randomly rough surfaces. The surfaces are invariant in one direction and the Dirichlet (pressure release) bound-

ary condition was used. The scattered field can be found exactly by solving an integral equation for the normal derivative of the pressure on the surface. This was done two ways by using linear integral equations of both the first and second kinds; the solutions were found to agree. Comparisons of the exact surface and scattered fields with those obtained using the Kirchhoff approximation were made. Significant differences in forward scattering were found between the Kirchhoff and exact results. This was true even when the Kirchhoff validity criterion, based on the radius of curvature, was well satisfied. Evidence will be presented to show that these differences are due to multiple scattering. [Work supported by ONR.]

10:35

**CC9. Equivalence of perturbation theories for sound scattering by rough surfaces.** Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery Science, University of Washington, 1013 N.E. 40th Street, Seattle, WA 98105)

Some approaches to rough-surface perturbation theory employ the Rayleigh hypothesis, but a method based on the extinction theorem avoids this assumption and gives a series which is apparently different. One might expect the latter method to be superior, but Winebrenner and Ishimaru have shown numerically that the two series give equivalent results for scattering from sinusoidal surfaces [D. P. Winebrenner and A. Ishimaru, J. Acoust. Soc. Am. Suppl. 1 75, S73 (1984)]. We have shown that the two series are the same for arbitrary surfaces. This proof depends upon term-by-term reciprocity of the perturbation series.

10:50

**CC10. Numerical computation of pulse response functions in an oceanic environment with rough surface scattering.** Lan Nghiem-Phu and Frederick Tappert (Daubin Systems Corporation, 104 Crandon Boulevard, Suite 400, Key Biscayne, Miami, FL 33149 and Rosenstiel School of Marine and Atmospheric Science, Miami, FL 33149)

A recently developed split-step Fourier algorithm for solving the parabolic wave equation, exactly with the effects of rough surface scattering included [J. Acoust. Soc. Am. Suppl. 1 77, S102 (1985)], has been used to compute the pulse response functions in an oceanic environment where a surface duct is present, while the surface of the ocean is treated as a randomly rough pressure release boundary. In particular, we have studied the effects of surface roughness characterized by a typical  $k^{-3}$  spectrum parametrized by the rms roughness  $\sigma$  and correlation length  $L$  on the pulse response patterns. Coherency of the time domain transmission loss is examined as surface roughness is varied in a manner similar to that discussed by the authors in a study of rough ocean bottom scattering. [Symposium on Ocean-Seismo Acoustics, La Spezia, June 1985.]

11:05

**CC11. Nearfield scattered acoustic waves in shallow-water waveguides with boundary topography.** Jacques R. Chamuel (Sonoquest/Advanced Ultrasonics Research, P. O. Box 584, Sudbury, MA 01776) and Gary H. Brooke (Defense Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

Experimental results are presented demonstrating the importance of backscattered, forward-scattered, and locally reactive acoustic fields created by shallow-water acoustic waveguide boundary topography. In a perfect waveguide, the transmission of a pulse may create a set of nondispersed waveguide pulses equivalent to the summation of highly dispersive normal modes. The presence of a boundary scatterer acts as a new delayed sound source in the waveguide. This new source produces a second set of delayed waveguide pulses. Each of the topographic features generates a new set of waveguide pulses delayed accordingly. The resulting acoustic field in the waveguide is complex. A technique has been devised to separate the original flat-waveguide over a periodic region creates local standing-wave-like behavior in the nearfield of the scatterers. Although the source-receiver range may be great, the distance between the receiver and the topographic features is small. The importance of the inhomogeneous normal modes increases as the receiver is moved closer to the scatterer. The received signal is greatly influenced by the relative position of the

rough waveguide region with respect to both source and receiver. [Work supported by DREP.]

11:20

**CC12. Near grazing forward scatter over a low roughness rigid ocean bottom.** Herman Medwin<sup>a)</sup> and Gerald L. D'Spain<sup>a)</sup> (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

When low-frequency sound interacts with a low roughness rigid surface at grazing or near-grazing angles, a significant coherent boundary wave is generated by repeated forward scatter. The boundary wave extends several wavelengths into the fluid and affects both the phase velocity and the amplitude of the forward propagating sound near the surface. The propagation is subsonic and slightly dispersive and, under certain conditions, the reflection coefficient can be greater than unity. Laboratory model results have not been extended to quantify these boundary wave phenomena in terms of the average height and rms slope of steep-sloped randomly rough surfaces. Predictions are made of the consequences of boundary wave propagation at a rough rigid ocean bottom. [Research supported by the Office of Naval Research.] <sup>a)</sup>Ocean Acoustics Associates, Pebble Beach, CA 93953.

11:35

**CC13. High-frequency bottom backscattering in the Arafura Sea.** Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery

Science, University of Washington, 1013 N.E. 40th Street, Seattle, WA 98105) and Michael D. Richardson (Naval Ocean Research and Development Activity, NSTL Station, MS 39529, and SACLANT ASW Research Centre, Viale San Bartolomeo 400, 19026 La Spezia, Italy)

Detailed backscatter and geoaoustic measurements were made at a 1-km<sup>2</sup> site in the Arafura Sea, north of Australia. Data were collected in collaboration with the Royal Australian Navy Research Laboratory in May 1984. Sediment geoaoustic and roughness properties were characterized using box core samples, underwater television, stereophotography, and sidescan sonar. Sediments consisted of a silty-clay matrix, mixed with up to 30% whole and broken sand and gravel-sized molluscan shells. X-radiographs of the sediment showed fine scale volume inhomogeneities due to burrowing animals and shell fragments. Compressional wave attenuation (65 dB/m at 125 kHz; CV = 47.5%) was much more variable than the sediment/water sound velocity ratio (0.989 at 125 kHz; CV = 0.66%). Backscatter measurements were made over the frequency range of 15–45 kHz and over a wide range of grazing angles. Scattering strength at a 20°-grazing angle (–27 dB) was independent of frequency. Comparison of measured results with those predicted by physical scattering models suggests that backscattering was primarily due to sediment inhomogeneity. Measurements obtained along a 144-km track showed that this scattering process was remarkably constant over a wide geographic area. [Work supported by NAVSEA 63-R and administered by NORAD.]

THURSDAY MORNING, 7 NOVEMBER 1985

SUITE 4, 9:00 A.M. TO 12 NOON

## Session DD. Architectural Acoustics III: Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture

Alfred C. C. Warnock, Chairman

*Division of Building Research, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada*

### *Invited Paper*

9:00

**DD1. The unreasonable effectiveness of number theory in acoustics.** Manfred R. Schroeder (Drittes Physikalisches Institut, Universitaet Goettingen, West Germany and AT&T Bell Laboratories, Murray Hill, NJ 07974)

Some of the problems that have occupied Vern O. Knudsen during his long career in acoustics have found surprising answers through applications of number theory, a field usually considered rather abstract even *within* mathematics. Among these unexpected applications are the distribution of normal modes in rooms, the design of surfaces ("reflection phase gratings") that optimally diffuse sound, and the construction of new music scales. Another application is the selection of phase angles to minimize the peak factor in waveforms for speech synthesis and sonar. Finally, precision measurements of reverberation and absorption of sound in gases, pioneered by Knudsen fifty years ago [V. O. Knudsen, *J. Acoust. Soc. Am.* 6, 199–204 (1935)], can be made more robust by using "pings" shaped according to the theory of finite number fields.

9:55–10:00

Break

### *Contributed Papers*

10:00

**DD2. Prediction of sound transmission through double-walled systems containing helium.** P. K. Raju and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36830)

Statistical energy analysis (SEA) is applied to study the transmission of random incidence sound waves through two finite independent panels separated by a gap filled with helium. The study was made for the following two cases: (1) glass panes with a cavity, and (2) vinyl panes with a

cavity. The analytical SEA model chosen is similar to the one developed earlier by one of the authors [A. J. Price and M. J. Crocker, *J. Acoust. Soc. Am.* **47**, 683–693 (1970)]. The analytical model consists of five linearly coupled oscillator systems, room–panel–cavity–panel–room. Both resonant and nonresonant transmissions have been considered for each wall of the double panel system. Allowance has also been made in the analytical model to include the effects of the absorption in the cavity. The results show that the use of helium instead of air in the gap of the double panel systems shifts the coincidence frequency region to a higher frequency and also increases the transmission loss values.

10:15

**DD3. Single-microphone method for certain applications of the sound intensity technique.** W. T. Chu (National Research Council of Canada, Division of Building Research, Ottawa, Ontario, Canada K1A 0R6)

Recent developments of the two-microphone intensity technique offer the possibility of improving measurements of acoustic impedance in impedance tube, sound transmission loss, and sound absorption coefficient in reverberation room. In these applications, the cross spectrum  $G_{p,p_1}$  of the two microphone signals taken simultaneously is required. However, with the application of a deterministic source signal such as the periodic pseudorandom sequence [W. T. Chu, *J. Acoust. Soc. Am.* **76**, 475–478 (1984)],  $G_{p,p_1}$  can be obtained using a single microphone to measure the pressure at the two locations sequentially. Some of the inherent difficulties associated with the two microphone method are thus eliminated. The proposed method and some of the preliminary results will be presented.

10:30

**DD4. Normal incidence absorption properties of single layers of elastic porous materials.** J. S. Bolton (Ray W. Herrick Laboratories, Purdue University, West Lafayette, IN 47907)

Recently, a theory has been developed which describes wave propagation in relatively stiff, partially reticulated polyurethane foams, the type most commonly used in noise control applications [J. S. Bolton and E. Gold, *J. Acoust. Soc. Am. Suppl.* **1** **77**, S59 (1985)]. A high impedance wave associated with the bulk mechanical properties of the foam matrix is usually significantly excited in these materials. As a consequence, the acoustical performance of finite depth layers of foam of this type is very sensitive to the boundary conditions which apply at the front and rear layer surfaces. Specifically, it will be shown in this paper that the action of a film facing is dependent on how it is attached to the foam layer. In addition it will be demonstrated that a small gap, e.g., 1 mm, separating a foam layer from a hard backing can increase the low-frequency absorption dramatically. A similar effect occurs when a film facing is not bonded directly to the surface of a foam layer but is separated from it by a thin air gap. This work has suggested an arrangement for enhancing the low-frequency absorption of thin foam layers.

10:45

**DD5. The “seat-dip phenomenon”: Low-frequency thief plagues concert goers.** David A. Greenberg (Acoustics Program, Applied Science Building, Pennsylvania State University, University Park, PA 16802)

Sound propagating at grazing incidence over auditorium seats is known to exhibit anomalous attenuation (as much as 20 dB) at low frequencies. This so-called “seat-dip phenomenon,” is predominantly a quarter-wave resonance associated with the height of the seat backs and is on the order of  $1\frac{1}{2}$  oct wide, centered at about 150 Hz [T. J. Schultz and B. G. Watters, *J. Acoust. Soc. Am.* **36**, 885–896 (1964) and G. M. Sessler and J. E. West, *J. Acoust. Soc. Am.* **36**, 1725–1732 (1964)]. This frequency range happens to correspond to an orchestra’s low end (bassoon, horn, trombone, tuba, cello, and string bass) and so the control of the seat dip is an important factor in the communication of orchestral balance. Toward this end, a simple scale model was built to investigate this complex sound field, particularly that between the seats. Various model configurations were utilized to investigate the sound pressure and intensity fields.

11:00

**DD6. Two-ear correlation in statistical sound fields.** Ian M. Lindevald and A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

The normalized correlation function  $R(f)$ , connecting the signals at the left and right ears, was measured in a reverberant room as a function of frequency. Head diffraction effects are made evident by comparing these data to the correlation function predicted and observed for two mikes separated by a headwidth (15 cm) of space and moving in the same room. In contrast to the familiar two-mike sinc ( $\delta$ ) correlation ( $\delta = kd$ ), head diffraction gives a two-ear correlation function, which is well described by  $\text{sinc}(\alpha\delta)/[1 + (\beta\delta)^4]^{1/2}$ , where  $\alpha = 2.2$  and  $\beta = 0.5$ . The dominant effect of head diffraction is to lower  $f_D$ , the decoupling frequency [first zero of  $R(f)$ ], to 500 Hz, about one-half the no-head value. The denominator in the fitted correlation function represents the near total independence of the signals at the two ears above  $f_D$ . Thus, with head diffraction, the variance in a paired-mike room with averaged pressure amplitude estimate at around 500 Hz, is already as small as that obtained sans head only at frequencies above 1200 Hz. [Work supported by NSF.]

11:15

**DD7. The RASTI method, objective measurements of speech intelligibility.** Klaus Højbjerg [Brüel & Kjær, Nærum Hovedgade 18, DK-2850, Denmark]

In recent years, there has been increasing interest in objective measurements of speech intelligibility. For rooms with long reverberation time (also those in which sound systems have been installed), it is very useful for control purposes to have a fast objective way of measuring speech intelligibility. RASTI (rapid speech transmission index) is a new standardized method of measuring speech intelligibility, which provides a rating in less than 10 s. The theory and principles of operation will be presented. Measurement results from the “Grundtvigskirken,” which is a Danish church having a very long reverberation time and a recently installed sound system, will be reported.

11:30

**DD8. Comparison of amplification systems in a small and large room with normal-hearing and special listeners.** Anna K. Nábelek and Amy M. Donahue (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740)

Listeners with either hearing loss or perceptual problems require a good speech-to-noise ratio and short reverberation for their best speech discrimination. These conditions are sometimes difficult to achieve. Assistive listening systems, in which signal is transmitted by infrared light (IR) or modulated radio waves (FM) are promoted for the special listeners. We compared the effectiveness of assistive systems in a classroom and in a large auditorium with several groups of listeners: normal hearing, hearing impaired, hearing aid users, elderly, and nonnative. In both rooms, the IR and FM systems allowed for better speech perception scores than through loudspeakers. The difference between the two assistive systems was very small. While all groups performed better through the assistive listening systems, the groups who achieved the greatest advantage had the poorest performance in the sound field. Since both systems were equally effective in improving speech intelligibility in less than optimal listening conditions, selection of a system should be based on such factors as cost, maintenance, and expected interferences. [Work supported by NIH.]

11:45

**DD9. Three-dimensional audio image reconstruction.** Anthony J. Romano and Jiri Tichy (Graduate School in Acoustics, Pennsylvania State University, State College, PA 16804)

An imaging technique is developed for the three-dimensional reconstruction of a desired sound field within a playback environment.



Through proper coupling of room design and speaker placement, and deduction of the needed inputs to the system to obtain the desired outputs (for a finite length time window of data), a discrete-sliding window algorithm is implemented to obtain the necessary time-domain inputs. As a

result of these procedures, the feedback/reverberatory effects of the playback environment are annihilated, leaving any desired sound field, while maintaining the three-dimensionality of the original field.

THURSDAY MORNING, 7 NOVEMBER 1985

DAVIDSON ROOM B, 9:00 TO 11:45 A.M.

### Session EE. Physical Acoustics VI: Low-Temperature Acoustics

Julian D. Maynard, Jr., Chairman

Department of Physics, Pennsylvania State University, 104 Davey Laboratory, University Park, Pennsylvania 16802

#### Invited Papers

9:00

**EE1. A review of the acoustic properties of the Bose liquid  $^4\text{He}$  and the Fermi liquid  $^3\text{He}$ .** Isadore Rudnick (Physics Department, University of California, Los Angeles, CA 90024)

Between the temperatures of 2.2° and 5.2° K,  $^4\text{He}$  is an ordinary liquid and the sound which occurs in it is ordinary sound. At 2.2° K the lambda phase transition occurs and below  $T_\lambda$  as a result of the Bose-Einstein condensation  $^4\text{He}$  is a superfluid with a galaxy of sound modes; first through fifth. Superfluidity and the acoustic modes persist as  $T \rightarrow 0$ . Their properties and use in determining the superfluid properties of  $^4\text{He}$  will be discussed. The critical point of  $^3\text{He}$  is 3.3° K. It too, undergoes a superfluid transition but at much lower temperatures—at about  $2 \times 10^{-3}$ ° K. Both above and below this temperature there is a new sound mode called zero sound. It exists both as a transverse and a longitudinal wave. Evidence that this mode arises because  $^3\text{He}$  is a viscoelastic liquid will be presented. Zero, first, second, and fourth sound have been observed in the superfluid phases of  $^3\text{He}$ . [Work supported by ONR.]

9:30

**EE2. Nonlinear acoustics at low temperature.** S. Putterman (Physics Department, University of California, Los Angeles, CA 90024)

The anomalous dispersion, strong nonlinearity, and small transport coefficients of  $\text{He}^4$  make it an ideal medium for establishing the frontiers of nonlinear acoustics. These properties make it possible for  $\text{He}^4$  to exhibit a well-defined three-wave interaction, where the high-frequency wave is analogous to the pump in a parametric amplifier (the low-frequency waves are the signal and idler). In fact, a macroscopic coherent sound wave will decay dramatically fast due to the parametric amplification of its first phonon of spontaneous decay [J. S. Foster and S. Putterman, *Phys. Rev. Lett.* **54**, 1810 (1985)]. It should be possible to observe similar effects with capillary waves (ripples). The properties of  $\text{He}^4$  as well as the roton minimum in its dispersion law also make it an ideal medium for searching for two- and three-dimensional localized states which are generalizations of the nonlinear Schrödinger equation and Kadomtsev-Petviashvili solitons. Finally,  $\text{He}^4$  should be used to probe the spectrum of acoustic turbulence. The self-similarity of the stationary turbulent state is a symmetry which will be broken quantum effects at high frequency.

10:00

**EE3. Surface acoustic wave investigation of superconducting films by means of the acoustoelectric effect.** Moises Levy (Physics Department, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

The piezoelectric interaction of surface acoustic waves (SAW) with metallic films deposited on a piezoelectric substrate is quenched when the film becomes superconducting. This acoustoelectric interaction should be proportional to the sheet resistivity of the film in the limit where the SAW period is smaller than the acoustoelectric relaxation time. The attenuation coefficient of SAW propagating through superconducting films of granular Pb and NbN has been measured at 700 MHz. In both cases the attenuation is not proportional to the measured dc resistivity of the films. The attenuation remains finite in the superconducting state of the granular Pb film even when the sheet resistivity disappears. This difference is ascribed, with a percolation model, to the fact that the dc current measures the resistance of the whole film while the SAW measures the effective resistance of a sample with dimensions comparable to its wavelength. The attenuation in the NbN film attains its normal value when the temperature increases slightly above the Kosterlitz Thouless transition where antiparallel flux line pairs dissociate, while at this temperature the film resistivity has barely changed from zero. A model has been proposed that ascribes the attenuation to the presence of normal cores in the flux line pairs. [Research supported by Air Force Office of Scientific Research under AFOSR Grant No. 84-0350.]

10:30

**EE4. Ultrasonic propagation in solid  $H_2$ .** Horst Meyer (Department of Physics, Duke University, Durham, NC 27706)

Besides the loss mechanism from imperfections and dislocations in a periodic lattice, an additional acoustic loss can be predicted from orientational ordering in this molecular quantum solid. Two modifications of  $H_2$ , namely ortho- $H_2$  (with rotational angular momentum  $J = 1$ ) and para- $H_2$  (with  $J = 0$ ) are randomly distributed in this solid, and we label by  $X$  the ortho- $H_2$  molefraction. As the temperature decreases, the ortho- $H_2$  molecules, which are coupled by anisotropic electrostatic forces, become orientationally ordered. This situation is similar to the ordering in diluted magnetic system with random distribution of spins. This talk will review the ordering in solid o- $H_2$  p- $H_2$  mixtures, the detection by acoustic and NMR techniques, and the correlation between these two methods. Furthermore, acoustic methods give a sensitive probe to the martensitic hcp→fcc transition that is induced by the ordering for  $X > 0.53$ . For lower  $X$ , the ordering takes place progressively in the hcp phase. [Work supported by NSF.]

11:00

**EE5. Ultrasonic studies of metallic glasses at low temperatures.** C. Elbaum (Brown University, Providence, RI 02906)

Amorphous materials, in general, including metallic glasses, exhibit very distinct characteristics at low temperatures ( $T \lesssim 2$  K), which are qualitatively quite different from those of their crystalline counterparts. Among these characteristics are unusual ultrasonic propagation features, such as a saturable, amplitude-dependent attenuation, and an "anomalous" temperature dependence of wave velocity. To account for these, as well as various thermodynamic and transport properties of amorphous solids, a phenomenological model developed some time ago, is often successfully used. This model postulates the existence of two-level systems (TLS) that give rise to excitations resulting from quantum mechanical tunneling of some unspecified entity (an atom or group of atoms) between the minima of an asymmetric double well potential. A brief review will be given of earlier work in this area followed by an account of recent ultrasonic studies conducted to probe the properties of metallic glasses, in general, and the features of the TLS model, in particular. Attention will be directed to the search for the specific nature of the as yet unknown tunneling entity of the model.

### Contributed Paper

11:30

**EE6. Freezing of liquids in small pores.** Kevin Warner and John Beamish (Department of Physics, University of Delaware, Newark, DE 19716)

We have ultrasonically studied the freezing of various cryogenic fluids in a porous solid. By measuring the transverse and longitudinal sound velocities for the empty matrix as well as for the cases of liquid- and solid-filled pores, we could determine both the density of the material in the

pores and its contribution to the composite material's elastic moduli. For fluid-filled pores, we find good agreement with the Biot model. Freezing in the pores is observed through changes in the sound velocities. In small pores, considerable super cooling of the liquid occurs and there is hysteresis between freezing and melting temperatures. Possible interpretations of these results in terms of interfacial free energies and wetting of the solid surface by the pore material are discussed. [Work supported by the University of Delaware Research Foundation and by the Research Corporation.]

THURSDAY MORNING, 7 NOVEMBER 1985

REGENCY BALLROOM II AND III, 9:00 A.M. TO 12 NOON

### Session FF. Psychological Acoustics III: Masking and Pitch

Daniel L. Weber, Chairman

*Department of Psychology, Wright State University, Dayton, Ohio 45435*

### Contributed Papers

9:00

**FF1. Two- versus four-tone masking, revisited.** Robert A. Lutfi (Waisman Center, University of Wisconsin, Madison, WI 53705)

Canahl [J. Acoust. Soc. Am. **50**, 471-474 (1971)] measured thresholds for a 1-kHz sinusoid masked either by two or by four surrounding tones. He reported four-tone masked thresholds that exceeded, by 5-7.5 dB, the

energy sum of the masking produced by the individual tone pairs. The present paper reports on a series of experiments investigating the effects of several factors on this 5-7.5 dB "excess" masking. In each experiment, thresholds for a 1-kHz, 250-ms sinusoid were measured as a function of the overall level of two or four equal amplitude sinusoids with frequencies arithmetically centered around 1.0 kHz. For conditions similar to those of the Canahl experiment, 5-6 dB of excess masking were obtained indepen-

dent of the level of the masking tones. Randomly varying overall level across presentations had no effect on the excess masking, nor did the frequency separation between tones in one condition. The excess masking was reduced or eliminated when the masking tones were generated using an amplitude modulation technique, when they were gated on and off with the signal, or when their waveshapes were fixed across trials. Canah's result may reflect listeners' ability to detect the signal as a variation in the waveshape of the multitone masker.

9:15

**FF2. A comparison of psychophysical tuning curves in simultaneous and forward masking.** Sid P. Bacon (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131 and Department of Experimental Psychology, University of Cambridge, Cambridge CB2 3EB, England) and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Cambridge CB2 3EB, England)

Threshold for a short-duration pure-tone signal depends upon the signal's temporal position within a longer duration pure-tone masker, particularly for masker frequencies above the signal frequency [Bacon and Viemeister, *J. Acoust. Soc. Am. Suppl.* 1 76, S75-76 (1984)]. We have extended the previous work to show how this temporal effect influences a common measure of frequency selectivity. Psychophysical tuning curves (PTCs) were measured for a 20-ms, 1-kHz signal presented at 10-dB sensation level and positioned at the beginning, at the center, at the end, or immediately following a 400-ms masker. In simultaneous masking, the PTCs were broadest for the signal at masker onset, and sharpest for the signal at the center; the differences were largest on the high-frequency side. For two of the three subjects, the forward masking PTC was sharper than any of the simultaneous masking PTCs. For one subject, however, there was little difference between the forward and the simultaneous masking PTC with the signal *temporally centered*. These results indicate that, at least for short-duration signals, the frequency selectivity measured with simultaneous masking PTCs, and the degree of sharpening revealed in forward masking PTCs, depend upon the temporal position of the signal within the simultaneous masker. [Work supported by NIH and MRC.]

9:30

**FF3. Effects of changes in absolute signal level on psychophysical tuning curves in quiet and noise in patas monkeys.** D. W. Smith, D. B. Moody, and W. C. Stebbins (Kresge Hearing Research Institute, University of Michigan Medical School, Ann Arbor, MI 48109)

Forward masking PTCs were measured in patas monkeys (*Erythrocebus patas*) at 500 Hz and 2, 4, and 8 kHz and signal levels of 10, 30, and 60 dB SL in quiet, and at 10 dB above masked threshold in two levels of wideband noise. Noise levels were sufficient to increase masked test-tone thresholds to levels that approximated the absolute signal levels tested at 30 and 60 dB SL in quiet. The background masker was used to control for "off-frequency" listening, which can act to broaden PTCs at higher signal levels, by maintaining a constant signal-to-noise ratio with increases in SPL. Results in quiet are in agreement with those reported in the human literature demonstrating broadening of the PTC as signal level is increased. The PTCs measured in noise also demonstrate a similar broadening, or loss of selectivity, at higher SPLs. These findings are not in agreement with a previous study [Green *et al.*, *J. Acoust. Soc. Am.* 69, 1758-1762 (1981)] that attempted to control off-frequency listening with background maskers at different signal intensities in humans. Differences in findings might be explained as differences in acoustic or experimental parameters between studies. [Research supported by NIH Grant No. NS 05785.]

9:45

**FF4. Peripheral and central components of comodulation masking release.** Mark Haggard, Anthony D. G. Harvey, and Robert

P. Carlyon (MRC Institute of Hearing Research, Nottingham University, Nottingham NG7 2RD, United Kingdom)

Comodulation masking release (CMR) is the improvement in threshold that occurs when the subaudio amplitude modulations of a masker envelope in the critical band around a target tone are replicated on one or more flanking bands outside the critical band; the phenomenon is of obvious significance for perceptual grouping and attention and may stimulate a literature on parametric variations as large as that for the BMLD. Basic general questions first need to be answered about the nature of the effect, specifically concerning the relative roles of (presumably peripheral) psychophysical suppression and of (presumably central) temporal pattern comparison. In the first experiment the effect was tested monotonically, diotically, and dichotically with one monotic control condition implementing 40-dB intensity disparity (as might result from worst-case transcranial conduction in the dichotic condition). Although intensity effects were present, a significant CMR was observed dichotically, suggesting that peripheral explanations such as suppression must be incomplete. The second experiment varied the phase (0°, 90°, 180°, and 270°) of the critical band and flanking band modulations. This experiment also produced results that cannot be completely explained by either a central statistical correction model, or a peripheral suppression model alone; negative CMRs were seen in some subjects.

10:00

**FF5. Forward masking of single and double component probes.** Gregory J. Fleet and John C. Booth (Departments of Psychology and Communicative Disorders, University of Western Ontario, London, Ontario, Canada N6A 5C2)

A series of preliminary studies sought to examine the role of temporal information in auditory frequency resolution. A forward masking paradigm was used, similar to that used in measuring psychophysical tuning curves. The masker levels were adapted to just mask a brief (10 ms) low-level signal. In one condition, though, a second signal, presented at near threshold, was added. Therefore, the signal consisted of two frequency components presented simultaneously following the masker. The masker frequencies were 1.0, 1.1, 1.2, 1.3, 1.4, and 1.5 kHz. Signal frequencies were 1.0 and 1.5 kHz. The level of the signal components were combinations of 10, 5, and 0 dB SL. The results for the 10/10-dB and 5/10-dB signal combinations were predictable from the single frequency signal studies, but greater masker levels than expected were found in the 0/10-dB SL conditions. Therefore, adding energy to the signal (at levels of "near" detectability), increased the masker-level thresholds much more than found in the single-signal condition. The discussion will focus on the type of information present in the task that could account for this effect. [Work supported by MRC.]

10:15

**FF6. Spectral locus and spectral spacing as determinants of the perceptual organization of complex tone sequences.** Punita Singh (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

The perceptual organization of complex tone sequences into subsequences or "streams" was studied using spectral differences between tones as a basis for eliciting segregation. The absolute position of the spectra was varied to provide timbral differences between tones, while relative spacing between harmonics was varied to provide differences in pitch. These attributes were put in competition in rapidly occurring sequences of the form:  $T_x P_x T_y P_x T_x P_y T_y P_y$ , with the first pair of tones being assigned the pitch  $P_x$  but different timbres  $T_x$  and  $T_y$ , and the second pair pitch  $P_y$ , and similarly different timbres. Six listeners indicated their percept of such sequences as being groupings based on timbral similarity, pitch proximity, or ambiguous patterns not dominated by either cue. The results demonstrated that the stream segregation phenomenon may be based on relative changes in spectral locus and spectral spacing of the sequential tones, and imply that timbre and pitch can both serve as cues, with tradeoffs between these attributes determining the perceptual organization of a complex auditory input. [Work supported by NINCDS.]

**FF7. Bandwidth, apparent pitch, and frequency discrimination: Effects with complex harmonic tones.** C. Douglas Creelman (Department of Psychology, University of Toronto, Toronto, Ontario, Canada M5S 1A1)

Harmonically complex signals which are heard as having low pitch ( $<0.3\text{--}0.5$  kHz) yield superior pitch discrimination compared to equivalent-pitch simple tones. On the other hand, narrow-bandwidth complexes, heard to have pitch more nearly that of a carrier frequency rather than the spacing of sidebands, are discriminated nearly as well as pure tones of equivalent pitch. Frequency discrimination was measured using the PEST adaptive procedure and small logarithmic steps for harmonic spectra with center frequencies in the range 240–4800 Hz, and fundamental frequencies from 60–300 Hz, covering much of the range of musical and voice formants. These data, and a further experiment in which frequency discrimination was measured between wide bandwidth complexes when the signals were of unpredictable amplitude, are considered to rule against a recently published model from Horst *et al.* [J. Acoust. Soc. Am. 76, 1067–1075 (1984)] which proposed amplitude differences among specific components to be the basis of frequency discrimination. Instead a profile analysis based on activity in the frequency range of cochlear sensitivity is proposed. [Work supported by the Canadian National Science and Engineering Research Council.]

10:45

**FF8. Interaural discrimination of phase and level differences of spectral sections of low-frequency noise.** William A. Yost (Parham Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

A 2000-Hz, low-passed noise was generated with a band of frequencies either interaurally phase shifted or presented with an interaural level difference. When an interaural level difference was generated, the level of all components at one ear was the same and a band of the noise was reduced in level at the other ear. The level of all the components was the same when an interaural phase shift was introduced for a band of the noise. When a band of frequencies are interaurally phase shifted, a pitch (sometimes referred to as the Huggins pitch) is produced. The just discriminable differences of interaural phase or level were measured as a function of the band width of the interaurally changed frequencies and the spectral location of the band. Listeners were more sensitive to interaural phase differences than to interaural level differences. These discriminations appear to be made on the basis of the pitch produced by the stimuli. The listeners report that when these pitches occur, the pitch is lateralized toward one side of the head while the noise timbre is lateralized on the opposite side. The results will be discussed in relation to the concept of a central spectrum and the mechanisms responsible for pitches similar to the Huggins pitch. [Research supported by grants from the NIH (NINCDS) and the AFOSR.]

11:00

**FF9. On the effect of amplitude envelope on the pitch of short complex tones.** T. D. Rossing (Physics Department, Northern Illinois University, DeKalb, IL 60115) and A. J. M. Houtsma (Institute for Perception Research, P. O. Box 513, 5600 MB Eindhoven, The Netherlands)

It has been shown that the perceived pitch of a short exponentially decaying sine tone is judged higher than the pitch of a gated sine tone of the same frequency and energy, that the frequency difference for equal pitch is adequately described by the relation  $\Delta f = K \log f$ , and that  $\Delta f$  grows monotonically both with exponential decay rate and with intensity [T. D. Rossing and A. J. M. Houtsma, J. Acoust. Soc. Am. Suppl. 1 77, S36 (1985)]. We have now extended the studies to complex tones consisting of two sinusoidal components with frequencies  $f_1 = mf + 50$  Hz and  $f_2 = (m + 1)f + 50$  Hz, where  $m$  equals 3, 4, or 5. Subjects were asked to judge the "low" pitch as well as the pitch of the individual spectral components  $f_1$  and  $f_2$ . We find shifts in  $f_1$  are generally less, and shifts in  $f_2$  are generally greater than the shifts observed in isolated sine-wave tones of comparable frequencies. Shifts in the perceived pitch of both components  $f_1$  and  $f_2$  show only slight dependence on the envelope decay rate in contrast to isolated sine-wave tones, where  $\Delta f$  increases monotonically with exponential decay rate. The "low" pitch of the complex is generally found to be 0–1.5% higher than that computed for the first-order pitch shift, and

it appears to depend less on envelope decay rate than the pitch of sine-wave tones of comparable frequency. [Work supported by a grant from the Netherlands Organization for the Advancement of Pure Research (ZWO).]

11:15

**FF10. Effects of removing the "spectral dominance region" upon noise repetition pitch.** James A. Bashford, Jr. and Richard M. Warren (Department of Psychology, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

The results of the present study indicate that the spectral dominance region is neither essential for accurate matching of co-phasic repetition pitch (RP +) nor the sole determinant of matches for antiphase repetition pitch (RP –). Listeners matched sinusoids to RP + and RP – at delays ( $t$ ) corresponding to repetition frequencies ( $1/t$ ) from 110 to 196 Hz. Stimuli were presented as broadband, low-pass filtered from the seventh spectral peak, high-pass filtered from the seventh peak, and notch filtered between the third and seventh peaks. Filtering slopes were 196 dB/oct with rejected components replaced by complementary Gaussian noise. Filtering had little effect upon pitch matching to  $1/t$  Hz for RP +. Depending upon filtering conditions, matches for RP – deviated from  $1/t$  Hz by about 5% to 10%. It is suggested that RP is based upon both the spacing of stimulation maxima ( $1/t$  Hz for RP + and RP –) and the loci of stimulation (which correspond to integral multiples of  $1/t$  Hz only for RP +). Pitches heard with RP – correspond to resolution of conflicts between loci and spacing, with different compromise pitches found for different spectral ranges.

11:30

**FF11. Timbre discrimination by 7-month-old infants.** Marsha G. Clarkson, Rachel K. Clifton, and Eve E. Perris (Psychology Department, University of Massachusetts, Amherst, MA 01003)

While previous research has demonstrated the infant's ability to categorize sounds according to their pitch in the presence of irrelevant spectral variations, little or no research has evaluated discrimination of those spectral variations. In the present study, 7- to 8-month-old infants demonstrated the ability to discriminate harmonic tonal complexes from two timbre categories that differed in their spectral energy distributions. These differences in timbre were identical to the irrelevant variations over which infants categorized sounds in earlier work. In a conditioned head-turning paradigm, infants initially learned to discriminate stimulus tokens that contained the same fundamental frequency, 200 Hz, but different harmonic components (e.g., harmonics 5–10 versus harmonics 3–8 or 7–12). After successfully completing the basic timbre discrimination, the same infants transferred their learning to missing fundamental stimuli that had novel spectral energy distributions (e.g., harmonics 4–9 versus harmonics 2–7 or 6–11). These results suggest that young infants can analyze the spectra of harmonic tonal complexes and discriminate differences in one of the most important cues for adults' timbre perception, the distribution of spectral energy. [Work supported by NSF No. BN583-04419 and NIMH No. MH00332.]

11:45

**FF12. Tonal implications of musical chords.** William F. Thompson and Lola L. Cuddy (Department of Psychology, Queen's University at Kingston, Kingston, Ontario, Canada K7L 3N6)

Two experiments examined the tonal implications of three different chord constructions. All chords were derived from the major triad in root position. Triadic chords contained all three notes of the triad; dyadic chords eliminated either the third or the fifth of the triad. In experiment I a triadic chord or dyadic chord was paired with a shepard tone. In experiment II a triadic chord was paired with either another triadic chord or a dyadic chord. In both experiments the second member of the pair was located at one of 12 randomly selected transpositions and listeners rated how well the second member of each pair followed the first. Results of the rating profiles bear on earlier work suggesting that different chord constructions emphasize different aspects of musical key relationships, information that may influence perceived tonal movement.

**Meeting of Accredited Standards Committee S12 on Noise**

to be held jointly with the

**Technical Advisory Group for ISO/TC 43/SC1 Noise**

K. M. Eldred, Chairman S12

*P. O. Box 1037, Concord, Massachusetts 01742*

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC1

*Director, Biodynamics & Bioengineering Division, AFAMRL/BB U.S. Air Force, Wright-Patterson AFB, Dayton, Ohio 45433*

Working group chairpersons will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC1 activities will be discussed, and there will be a report on the meeting of ISO/TC 43/SC1 held in April 1985 in Budapest, Hungary.

THURSDAY AFTERNOON, 7 NOVEMBER 1985

SUITE 6, 1:30 P.M.

**Joint Meeting of Accredited Standards Committees S1 and S3**

The activities of S1 will be discussed first, proceeding to matters of interest to both S1 and S3, and concluding with S3 activities.

**Meeting of Accredited Standards Committee S1 on Acoustics**

E. H. Toothman, Chairman S1

*Bethlehem Steel Corporation, Room B-238, Martin Tower, Bethlehem, Pennsylvania 18016*

**Standards Committee S1, Acoustics.** Working group chairpersons will report on their progress in the preparation of standards, 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.

**Meeting of Accredited Standards Committee S3 on Bioacoustics**

L. A. Wilber, Chairman S3

*422 Skokie Boulevard, Wilmette, Illinois 60091*

**Standards Committees S3, Bioacoustics.** The current status of standards under preparation will be discussed. In addition to those topics of interest including hearing conversation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years.

The international activities in ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, for which S1 and S3 serve as the U. S. Technical Advisory Groups, will be discussed. There will also be reports on the meetings of ISO/TC 43 and IEC/TC 29 held in Budapest, Hungary, in April 1985.

**Session GG. Physiological Acoustics I: Chemical Composition and Electrophysiology**

Jon H. Kaas, Chairman

*Department of Psychology, Vanderbilt University, Nashville, Tennessee 37240***Contributed Papers**

1:30

**GG1. Tectorial membrane—A collagen-based gel?** Isolde Thalmann, Gertraud Thallinger, Thomas H. Comegys, and Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

According to traditional thinking, the tectorial membrane (TM) is a gel with a matrix of proteins, but containing no collagen. We recently reported patterns of amino acids and proteins of the TM consistent with the presence of substantial amounts of collagen [I. Thalmann *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 77, S93 (1985)]. Using two-dimensional polyacrylamide gel electrophoresis with extended alkaline range, we have now found that the predominant protein of the TM of the guinea pig virtually superimposes with purified collagen standards. Moreover, peptide mapping of cyanogen bromide digests ("fingerprinting") results in a pattern closely resembling that of collagen type II. Further procedures for identification and characterization of this protein (effect of collagenase, immunoblotting, etc.) are in progress. At this stage it is not possible to decide to what extent the amino and neutral sugars present in hydrolysates of the TM are part of the alleged collagen molecule. The cupula ampullaris, a vestibular superstructure thought to be analogous to the TM, exhibits a substantially different chemical profile. The functional significance of the findings will be discussed. [Supported in part by NIH.]

1:45

**GG2. Direct measurement of longitudinal movement of tracer in cochlear endolymph.** Alec N. Salt, Ruediger Thalmann, Daniel C. Marcus, and Barbara A. Bohne (Department of Otolaryngology, Washington University School of Medicine, St. Louis, MO 63110)

On the basis of morphological studies which show that tracers injected into cochlear endolymph subsequently reach the endolymphatic sac, it has been repeatedly postulated that there is a bulk flow of cochlear endolymph towards the basal turn. However, no study has yet reported the relative contributions of passive diffusion and volume flow to the dispersion of substances in endolymph. We have measured the characteristics of movement of an ionic tracer between the first and second turns of the guinea pig cochlea. Tetramethylammonium (TMA) was chosen as a tracer as it can be readily detected in low concentrations by a potassium-selective microelectrode. The characteristics of TMA arrival monitored in turn II following a minute injection into turn I were compared with those monitored in turn I following injection into turn II. These results were compared with those predicted by a one-dimensional computer model of ionic diffusion in which various rates of volume flow could be incorporated. Our results demonstrate that TMA appears to spread symmetrically in the cochlea. This finding demonstrates that the movement of ions in endolymph is dominated by passive diffusion with almost negligible contribution from bulk flow. [This work supported by NIH.]

2:00

**GG3. Intracellular recordings from supporting cells in the mammalian cochlea.** Elizabeth Desterle and Peter Dallos (Auditory Physiology Laboratories and Department of Neurobiology and Physiology, Northwestern University, Evanston, IL 60201)

The mammalian organ of Corti is constituted of a variety of different cell types. We have utilized horseradish peroxidase marking, along with

intracellular recording techniques, to examine whether this heterogeneity is also reflected in the cells' electrical properties. Recordings from third and fourth turn supporting cells, hair cells, and fluid spaces have been made from guinea pig cochleas. A fast Fourier analysis was performed on responses to tone bursts. In all supporting cell types examined, the magnitude of the ac response component recorded from within the supporting cells was less than or equal to that recorded from adjacent fluid spaces. In contrast, differences were observed with respect to the dc response component. Dc components greater than those recorded from adjacent fluid spaces were frequently seen in supporting cell types located in the inner hair cell region. A slowly increasing dc component reminiscent of the slow electrical potential changes that occur in glial cells was also seen. These slow, depolarizing potentials are similar to those recorded from goldfish saccular supporting cells [T. Furukawa, *J. Physiol.* in press]. [Work supported by NINCDS Grant NS-08635.]

2:15

**GG4. Abstract withdrawn.**

2:30

**GG5. The origins of adaptation in the auditory nerve.** Robert L. Smith (Institute for Sensory Research, Syracuse University, Syracuse, NY 13210) and Larry A. Westerman (Optical Radiation Corporation, 1300 Optical Drive, Azusa, CA 91072)

In response to stimuli of constant sound intensity, auditory-nerve firing rates are maximum at onset, and then decay or adapt towards a steady-state rate. The early decay appears to be made up of two exponential components, rapid adaptation and short-term adaptation, with time constants on the order of 5 and 50 ms, respectively. A variety of data will be presented and reviewed that generally, albeit indirectly, support the conclusion that both phases of adaptation have a common, synaptic, origin. The data demonstrate the following: adaptation is not present in the hair cell response, adaptation compresses the input-output characteristic, adaptation has an asymmetrical effect on responses to changes in intensity, adaptation does not qualitatively alter response variability, adaptation is minimally affected by stimulus rise time, adaptation is not caused by neural refractoriness, and adaptation is a complex function of sound frequency. A previously described [R. L. Smith and M. L. Brachman, *Biol.*

2:45

**GG6. Synchrony and "synchrony suppression" in primary auditory neurons.** Donald D. Greenwood (Audiology and Speech Sciences, University of British Columbia, Vancouver, BC V6T 1W5, Canada)

Synchrony of discharge to two-tone stimuli and "synchrony suppression" are analyzed by examining the implications of the definition of vector strength. "Synchrony suppression," defined as the reduction in the vector strength for one tone when a second is added, occurs by definition when half-wave rectification occurs in an otherwise linear system. The usual shifts of empirical vector strength curves occur, disproving the necessity for compressive nonlinearity. "Synchrony suppression," has been defined incompatibly as the shift in dB of a vector strength curve—said to be the magnitude of suppression. The identification of half-wave rectification with vector strength reduction disproves this conception, which is a logical fallacy as well. Curve shift is here defined as the shift of the crossover point at which a vector strength growth curve intersects the paired decay curve of the fixed-level tone. Crossover is the point of equality of the output amplitudes, hence vector strengths, in the period histogram. When the vector strength definition is applied to a complex waveform at the output of a compressive nonlinearity that compresses equal inputs equally, the shifts of the crossover points necessarily equal those in the linear case. But differences in unequal inputs will be accentuated in the relative output levels in the histogram, leading to greater differences of the vector strengths, at those relative input levels, than in the linear case. More visible effects of compression result from waveform distortion, which reduces vector strength saturation, and crossover, values and causes them to recede at higher levels. If the auditory periphery includes nonlinearities that compress *equal* inputs unequally, the basic curve shifts caused by half-wave rectification will change by the amount of differential compression. Since data suggest such nonlinearities may exist, it could be useful in new studies to use the crossover point of the growth and decay curves, as an index of equal output, to seek their confirmation and to infer something about them. [Work supported by NSERC.]

3:00

**GG7. Tuning in goldfish auditory nerve fibers.** Richard R. Fay and Timothy J. Ream (Parmly Hearing Institute, Loyola University of Chicago, 6525 N. Sheridan Road, Chicago, IL 60626)

Tuning curves for over 500 saccular nerve fibers were systematically surveyed with 10-Hz resolution between 100 and 2000 Hz using statistically defined spike rate thresholds with an automated tracking procedure. Spontaneous rate (SR), variation in rate, spontaneous type (silent, burst, irregular, and regular), and position of the electrode in the nerve were recorded. Best frequency (BF), best sensitivity (BS),  $Q$  10 dB, and tuning slopes were measured. Units vary considerably in BF and in degree of tuning at a given BF. Four main types of units were identified: BF at or below 150 Hz, BF between 160 and 390 Hz, tuned between 410 and 800 Hz, and tuned between 900 and 1700 Hz. Additional units were found having no BF (essentially flat, or having low-pass characteristics without measurable BF). BS varies over 60 dB with the most sensitive units within 10 dB of behavioral thresholds. The least sensitive units within each type had no spontaneous activity. Within each type, SRs range from zero to over 100 S/S. Average  $Q$  10 dB is 0.75 ranging from 0 to 2. Although units can be selected to illustrate a continuous range of BFs from 100 to 1700 Hz, the units clearly group into the above BF types which are also characterized by different adaptation patterns. In the ventral half of the nerve, fibers with the highest BFs were found medially, and those with the lowest BFs found laterally. [Supported by the NIH.]

3:15

**GG8. A comparison of AP and ABR tuning characteristics in the guinea pig.** Carolyn J. Brown and Paul J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52240)

AP and ABR tuning curves were measured using a forward-masking paradigm in guinea pigs with chronically implanted electrodes. Measurements were made before exposure to wide-band noise and at several intervals after exposure. The noise exposure was sufficient to produce temporary threshold shifts up to 60 dB lasting several days and resulted in a systematic widening of both AP and ABR tuning curves and a reduction in tip-to-tail ratios. No significant differences were found between simultaneously recorded AP and ABR tuning curves as measured by  $Q_{10}$ , tip-to-tail ratio, or the slope of the tuning curves. Tuning curves obtained by measuring AP and ABR response latency as a function of masker level, showed similar form and changes with hearing loss to those based on amplitude of response. The similarities between AP and ABR tuning characteristics provide evidence that the ABR is sensitive enough to peripheral changes to be a useful tool in the study of auditory frequency selectivity.

3:30

**GG9. Variations in ABR morphology, absolute latencies, and interpeak intervals due to electrical stimulation in man and guinea pig.** D. D. Brown and R. T. Miyamoto (Department of Otolaryngology—Head and Neck Surgery, Riley Hospital A56, Indiana University School of Medicine, Indianapolis, IN 46223)

Electrical auditory brainstem response (EABR) recordings have been made in 16 human subjects implanted with a single channel intracochlear device, and in 25 guinea pigs (GPs) with a similar intracochlear electrode. Electrical stimulation results in an ABR that differs from its acoustical counterpart. The most prominent change in the EABR is a decrease in absolute latency of all waves amounting to 1.0–1.5 ms in man and 0.6–0.7 ms in the GP. Not all waves have the same latency decrease with electrical stimulation, and this results in a change of some interpeak intervals. This change is more prominent in man (where it affects the I–III intervals) than it is in the GP. The EABR and acoustic morphologies differ; in man this difference is small while in the GP it is quite pronounced. Normative data will be presented and the differences found between electrical and acoustical stimulation will be discussed.

3:45

**GG10. Response characteristics and connections of auditory cortex in squirrels.** L. E. Luethke, L. Krubitzer, and J. H. Kaas (Department of Hearing and Speech Sciences and Department of Psychology, Vanderbilt University, Nashville, TN 37240)

While several subdivisions of auditory cortex have been determined in cats and monkeys, the organization of auditory cortex in rodents is not well understood. Connections, myeloarchitecture, and response characteristics of clusters of neurons were used to subdivide auditory cortex in grey squirrels, which are rodents with large, well-differentiated brains. These data reveal that primary auditory cortex (AI) is tonotopically organized (high frequencies represented caudomedially, low rostrrolaterally), heavily myelinated, and reciprocally connected with multiple locations in the surrounding cortex. Although many neuronal clusters outside of AI are broadly tuned and habituate rapidly, our findings show that some of the surrounding cortex is responsive to auditory stimulation and includes two zones: (1) a rostrrolateral (to AI) region which appears to be tonotopically organized and (2) a newly discovered "parietal ventral" somatosensory field which is apparently bimodal. AI is also reciprocally connected with a caudolateral zone which is not responsive to auditory stimuli. These results suggest that auditory cortex in rodents is more complexly organized than previously thought. [Work supported by NIH Grant NS16446.]

## Session HH. Speech Communication V: Consonant Perception

Donald J. Sharf, Chairman

Speech and Hearing Sciences Program, University of Michigan, 1111 East Catherine Street, Ann Arbor, Michigan 48109

## Contributed Papers

1:30

**HH1. Syllable discrimination in 6–11 year-old normal and handicapped school children.** Lois L. Elliott, Randy Partridge, Jennifer Rupert, and Robert DeGraaff (Audiology and Hearing Impairment, Northwestern University, Evanston, IL 60201)

We have previously described age-related differences in VOT discrimination [L. L. Elliott *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 77, S96 (1985)] and discrimination of the place of articulation feature [L. L. Elliott *et al.*, *J. Acoust. Soc. Am.* 70, 669–677 (1981)] when just-noticeable differences among five-formant synthesized CV syllables are measured using adaptive test procedures. This is the first report of a larger study which also measured vocabulary level, receptive language skill, speech articulation, etc. As previously reported, older children demonstrated smaller just-noticeable differences for syllable discrimination than younger children. Of great interest, however, was the finding that, among normals, there were higher correlations between syllable discrimination and speech production for 6–8 year olds than for 8–11 year-old children. Preliminary factor analyses revealed different factor structures for learning-disabled 8–11 year olds, as compared to their normal age mates. For example, for these learning-disabled Ss, the two measures of VOT discrimination defined one factor while the two measures of the place features, plus the measure of receptive vocabulary, defined another factor. In contrast, the measures of syllable discrimination were distributed across four factors for the older, normal Ss. [Work supported, in part, by NINCDS, NIH.]

1:45

**HH2. Vowel constriction effects on the perception of F1 offset and vowel duration cues for word final stop consonant voicing.** Rebecca M. Fischer and Ralph N. Ohde (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232)

The perception of voicing in final velar stop consonants was investigated by systematically varying vowel duration and change in frequency of the final F1 transition. Several CVC continua were synthesized for each of three vowels, [i, ɪ, æ] which represent a range of tongue constriction values. Subjects used a scaling procedure in responding to these stimuli under both open and closed response set formats. The findings of this study show that both vowel duration and F1 offset cues influence the perception of final consonant voicing, with the latter more salient in open than constricted vowels and the former more important in constricted than open vowels. In addition, individual preferences for either vowel duration or F1 offset cues were observed, and these predilections were related to the vowel constriction feature. In summary, the results show that F1 offset cues were perceived more invariantly as + voice than vowel duration cues across vowel contexts.

2:00

**HH3. Effect of formant transition rate variation on the differentiation of synthesized child and adult /w/ and /r/ sounds.** Ralph N. Ohde (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232), and Donald J. Sharf (Speech and Hearing Sciences Program, University of Michigan, 1111 East Catherine Street, Ann Arbor, MI 48109)

Although a primary acoustic property differentiating /w/ and /r/ is the onset frequency of the second formant transition (F2) relative to the

onset of the third formant transition (F3), it has been hypothesized that the rate of change of F2 may provide additional cues to this glide distinction. In order to investigate this supposition, transition rate of F2 was systematically varied while holding F2 onsets constant for stimuli modeled after child and adult vocal tracts. Eight subjects each participated in the child and adult conditions. The results showed that rate significantly affected the perception of /r/ and /w/ for only the adult condition. However, the results were generally opposite hypothetical predictions. That is, as transition rate increased, /w/ identification significantly decreased. These findings are interpreted as supporting the perceptual salience of a slow transition rate for /w/, which is important in the glide-stop contrast in adult speech. Moreover, the results support the conclusion, that transition rate is not an important property in the differentiation of /r/ and /w/ relative to hypothetical predictions.

2:15

**HH4. Trading relations in the perception of /r/-/l/ by articulation-delayed children.** Shelina Manji, Winifred Strange, Linda Polka, and Michele Steffens (Department of Communicology, 255 CBA, University of South Florida, Tampa, FL 33620)

Articulation-delayed children who did not produce /r/ correctly and who showed slow progress in therapy, were tested on identification of two synthetic series contrasting prevocalic /r/ and /l/ in a trading relations paradigm. The primary spectral cue (F3 and F2 onset and transition) was varied in ten steps in both series. For one series, the secondary temporal cue (F1 steady-state and transition duration) was /r/-like; in the other, it was /l/-like. Result showed that phoneme boundaries along the spectral dimensions shifted appropriately as a function of the temporal cue value (i.e., a trading relation was demonstrated). However, the shift was somewhat smaller than for adults, and both boundaries were displaced toward the /l/ end of the spectral dimension, relative to the adults. These results suggest that such children have particular difficulty with the perceptual differentiation of spectral information, as might be predicted from their articulatory substitution patterns. [Supported by NIMH, NINCDS.]

2:30

**HH5. Reaction times to categorically perceived speech stimuli.** Donald J. Schum (Department of Speech Communication, 155 MDA, Louisiana State University, Baton Rouge, LA 70803), and Richard R. Hurtig (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

The measurement of reaction time for item identification has been suggested as one metric of the equivalence of category members [C. Mervis and E. Rosch, *Ann. Rev. Psychol.* 32, 89–115 (1981)]. Reaction times were measured while adult subjects ( $N = 10$ ) categorized computer-generated CV syllables. One set of tokens varied along the VOT continuum (/ga-ka/). A second set of tokens varied along the F2 starting frequency continuum (/ba-da-ga/). The subjects typically demonstrated increased mean reaction times for tokens at or near phoneme boundaries. In certain instances, tokens near a phoneme boundary were consistently identified as belonging to a certain phoneme category; however, significantly longer reaction times were required to make those decisions. This response pattern would suggest that all tokens within a given phoneme category may not be functionally equivalent. The results will be discussed in light of the perceptual demands of listening to ongoing speech.



**HH6. Japanese quail categorize /d/ across allophonic variation.** Keith R. Kluender, Randy L. Diehl (Department of Psychology, University of Texas Austin, TX 78712), and Peter R. Killeen (Department of Psychology, Arizona State University, Tempe, AZ 85287)

Studies using animal subjects provide a unique opportunity to assess the nature of processes for perceiving speech. Demonstrations that animals have the ability to categorize speech sounds in ways similar to humans, serve to undermine claims that some levels of speech processing either are specific to humans or carry substantial cognitive requirements. In our study, Japanese quail were taught to discriminate natural CV + /s/ syllables beginning with /d/ from those beginning with /b/ or /g/. The phoneme /d/ was chosen because of its well-documented context-conditioned variability. Birds received food reinforcement for pecking a lighted key during repeated presentation of /d/ syllables, but were required to refrain from pecking in order for the presentation of /b/ and /g/ syllables to be terminated. The quail were first trained to discriminate /daes/, /das/, /dus/, and /dis/ from tokens with the same vowels following /b/ and /g/. After reaching asymptotic performance for the /æ/, /a/, /u/, and /i/ contexts, birds were tested on novel tokens with the vowels /e/, /ɛ/, /v/, and /i/. They received no reinforcement for pecking to novel tokens. Quail pecked substantially more to novel /d/ tokens than to novel /b/ or /g/ tokens, suggesting that they treated the novel stimuli as similar to their non-novel counterparts. Apparently, there exist significant acoustic/auditory commonalities among the class of /d/ allophones that may serve as the basis of category formation for both humans and animals. [Work supported by NICHD.]

3:00

**HH7. Duration effects on labeling of tonal analogs to F1-cutback continua.** R. E. Pastore, C. Morris, and J. K. Layer (Department of Psychology, State University of New York at Binghamton, Binghamton, NY 13901)

F1 cutback has been demonstrated to be at least a contributing cue, and very possibly a major cue, for the perception of voicing contrasts. In an attempt to evaluate the feasibility of there existing an acoustic basis for the role of F1-cutback cues in the perception of voicing stimuli, our recent research has focused on comparing stimulus cue dependencies for voicing boundaries with similar stimulus changes in analogous tonal stimuli varying in complexity. An earlier paper [Pastore *et al.*, J. Acoust. Soc. Am. 75, S65 (1984)], described parallel labeling boundaries as a function of tonal analogs to F1 frequency and F2 transitions, but no stimulus duration. We now have identified conditions under which our F1-cutback analog stimuli exhibit both labeling boundary locations, and changes as a function of duration, which are characteristic of voicing boundaries for speech stimuli. [Supported in part by NSF.]

3:15

**HH8. Perception of sinewave analogs of stop consonant place information II.** James V. Ralston and James R. Sawusch (Department of Psychology, State University of New York at Buffalo, 4230 Ridge Lea Road, Amherst, NY 14226)

Identification and discrimination procedures were used to further explore a possible auditory basis for the categorical perception of place of articulation. Three 11-element stimulus series were constructed of either two or three pure tone components, with initial 40-ms frequency transitions followed by 200-ms steady-state portions. The initial transitions of the tonal stimuli mirrored the formant structure of synthetic two and three formant stop-place series varying in the onset frequencies of higher formant transitions. Subjects were operantly trained to classify stimuli into either two or three categories. Averaged ABX discrimination functions exhibited local maxima and minima similar to those found previously with analogous synthetic speech stimuli. These results suggest that a common auditory process contributes to discontinuities in the perception of both these tonal series and analogous stop-place series. The effects of stimulus structure on identification performance is also discussed. [Work supported by NINCDS.]

**HH9. The role of attention in selective adaptation of place.** John W. Mullennix (Department of Psychology, State University of New York at Buffalo, 4230 Ridge Lea Road, Amherst, NY 14226)

Selective adaptation of speech has been proposed by a number of researchers to be contingent upon a variety of differing mechanisms, including spectral overlap, phonetic featural overlap, and criterion changes. However, the role of attention has long been neglected within the adaptation process. In the present study, focused attention to an adapting syllable was manipulated. Subjects performed a dual-task dichotic monitoring procedure with the adaptor appearing either in an attended or unattended channel (ear). Categorization changes along a /bae/-/dae/ place-of-articulation continuum after adaptation were recorded. Also, the perceptual locus of effects (monaural or central) was assessed by the degree of the interaural transfer of adaptation. In a second and similar experiment, burst-cued adaptors varying in featural similarity to the /dae/ endpoint of the same continuum were utilized. The result of allocation of attentional resources to the adapting stimuli is discussed, along with the general relationship of attention to selective adaptation. [Work supported by NINCDS.]

3:45

**HH10. Time-intensity envelope cues for consonant recognition.** Sigfrid D. Soli, Virginia M. Kirby (Biosciences Laboratory, 270-4S-11, 3M Center, St. Paul, MN 55144), Dianne Van Tasell (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455), and Gregory P. Widin (Biosciences Laboratory, 270-4S-11, 3M Center, St. Paul, MN 55144)

A primary source of perceptual information for the cochlear implant user in the time-intensity envelope of the speech waveform. The purpose of this study was to estimate the amount and type of information for consonant recognition potentially available in the time-intensity envelope of speech. The experimental stimuli were generated from the time-intensity envelopes of 19 /aCa/ utterances (C = /p, t, k, b, d, g, f, θ, s, ʃ, v, ð, z, ʒ, m, n, r, l, j/), spoken by a male talker. The envelopes were obtained by full-wave rectifying and low-pass filtering the speech waveforms at 2000, 200, and 20 Hz. These envelopes were used to multiply noise with a 3-kHz bandwidth, producing three sets of envelope stimuli with identical, flat spectra that differed in the amount of time-varying in their amplitude envelopes. The unprocessed speech waveforms and the three sets of envelope stimuli were presented to 12 normal-hearing subjects in blocked, closed-set consonant recognition tests. Individual and group confusion matrices from each test were submitted to multidimensional scaling analyses (SINDSCAL) and information transmission analyses (SINFA). The results of these analyses revealed that three different envelope features account for most of the consonant information in the time-intensity envelope of speech.

4:00

**HH11. Mixed excitation for speech III: Effects of manipulation of low-versus high-frequency components on naturalness of voiced fricatives.** George D. Allen and Leah H. Jamieson (Department of Audiology and Speech Sciences and School of Electrical Engineering, Purdue University, West Lafayette, IN 47907)

In our previous presentation to the Society [J. Acoust. Soc. Am. Suppl. 1 77, S11 (1985)], we examined the preceptual effects of a variety of digital manipulations of naturally produced voiced fricatives /z/ and /ʒ/ in V-V contexts. Briefly, local reorderings of pitch periods within the fricative result in a "rough" perceptual quality, while repetitions of periods cause the fricative to sound "buzzy." In the present study, we examine in greater detail the source of these effects by comparing the relative contributions of low- versus high-frequency components of the pitch period to our perception. Each pitch period of the 24-kHz digitized waveforms was divided (by interactive hand marking) into an initial portion, consisting largely of low-frequency formant energy, and a final portion, consisting largely of high-frequency friction energy. The manipulations used in our previous study were then repeated on these subportions, leav-

ing the order of the other subportions intact. Preliminary results reveal a greater effect of the high-frequency component on the perceived distortions, but with variation from token to token and from talker to talker.

4:15

**HH12. Effects of listening to repeated syllables: Perstimulatory verbal transformations versus poststimulatory shifts in category boundaries.** Richard M. Warren and M. D. Meyers (Department of Psychology, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

Listening to repeated syllables produces two quite different effects depending upon whether observations are made following or during repetition. Poststimulatory changes in category boundaries (for example,

shifts in the voice-onset time boundary separation /tu/ from /du/ after hearing repetitions of either /tu/ or /du/) have been the subject of lively controversy concerning whether or not feature-detector adaptation (FDA) is responsible for such changes. Curiously, studies dealing with poststimulatory effects have not considered perstimulatory effects developing *during* stimulation. It is known that illusory changes (verbal transformations) occur while listening to repeated syllables, and the present study examined these verbal transformations for evidence of FDA. The repeated stimulus .../tu/, /du/, /tu/, /du/..., was used, since little or no net boundary change should be produced by FDA. The verbal transformation reported were inconsistent with adaptation theories. It is suggested that poststimulatory effects follow a "criterion shift rule" applying to perception in general, while verbal transformations tap special linguistic mechanisms employed normally to correct errors and enhance intelligibility of speech. [Work supported by NIH and NSF.]

THURSDAY AFTERNOON, 7 NOVEMBER 1985

SUITE 5, 1:30 TO 5:35 P.M.

## Session II. Underwater Acoustics V: Propagation and Inverse Methods

John E. Ffowcs-Williams, Chairman

*Engineering Department, Cambridge University, Trumpington Street, Cambridge CB2 1PZ, England*

Chairman's Introduction—1:30

### Contributed Papers

1:35

**III1. Model experiments on normal mode propagation in a wedge.** H. Hobaek, C. T. Tindle, and T. G. Muir (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

Experiments to measure normal mode propagation into and out of a shallow-water wedge have been conducted in an indoor tank. The wedge has a plane sand bottom and a slope which can be set up to 9°. A seven-element line source allows individual normal modes of low order to be excited. For downslope propagation, measured attenuation and group velocities are in agreement with simple adiabatic normal mode theory. Dispersion and attenuation are rapidly reduced with increasing slope. For downslope propagation, the usual vertical wavefronts associated with normal modes become curved into spheres centered on the source. Upslope propagation is dominated by the disappearance of modes as they reach their cutoff depth. The energy of a mode traveling upslope is rapidly lost to the bottom after cutoff with negligible coupling to lower modes. [Work supported by ONR.]

1:50

**II2. Experimental confirmation of horizontal refraction of sound propagation in a wedgelike ocean.** R. Doolittle, A. Tolstoy (Naval Research Laboratory, Code 5120, Washington, DC 20375-5000), and M. Buckingham (Royal Aircraft Establishment, Farnborough, Hampshire, GU146TD, United Kingdom)

Analysis of experimental data obtained in the region of the East Australian Continental Slope has confirmed theoretical predictions of energetic horizontal refraction due to sound-slope interaction. The observed modal behavior, azimuthal characteristics, and effects of bottom absorption were in accordance with both normal mode and ray theoretical solutions. The experiment, conducted with two ships (one towing a source and the other an array), started at a 400-m depth contour with an initial separation of 33 km and proceeded to deep water on a diverging but constant line of bearing course. Beamformed data-bearing shifts demonstrated the dominance of horizontally refracted arrivals for this geometry. A combi-

nation of source-array positions and critical angle effects served to limit both the azimuthal extent of sound received and the total number of modes arriving at any one position.

2:05

**II3. Temporal variability in shallow-water acoustic transmission and its correlation with the environment.** Hassan B. Ali (SACLANT ASW Research Centre, San Bartolomeo 400, I-19026 La Spezia, Italy)

The ocean is a complex, highly variable medium of acoustic propagation. Medium-induced variability in the acoustic index of refraction (sound speed) can have significant effects on acoustic propagation. Using the results of SACLANTCEN measurements in two diverse geographical areas, the present paper discusses the correlation between the temporal variability of ocean parameters and the fluctuations in acoustic propagation. Following a discussion of the relevant spatial and temporal scales, brief comments are made on the theoretical approaches to propagation in a random medium. This is followed by a discussion of selected results of measurements in the Mediterranean and in the North Atlantic. It is shown that fluctuations in acoustic transmission loss closely correlate with ocean variability resulting from internal waves, tides, tidally advected changes in water masses, and ocean fine structure.

2:20

**II4. Propagation of bottom interacting very low-frequency acoustic signals off the coast near Cape Fear, NC.** T. W. Tunnell and A. Kramer (Naval Ocean Research and Development Activity, Code 240, NSTL, MS 39529)

The Naval Ocean Research and Development Activity (NORDA) and the United States Geological Survey (USGS) at Woods Hole, MA, conducted the Cape Fear very-low-frequency (VLF, 20 Hz and less) exercise off the coast near Cape Fear, NC, during June 1985. The NORDA VLF array system (16-element vertical line array with a uniform spacing of 20 m) was deployed in 400 m of water in a bottom moored configuration. The

data acquired by the array system were collected by the R/VG. W. PIERCE via a radio link with the telemetry buoy on the surface attached to the array. The USNS LYNCH towed a continuous wave (cw, 10 Hz) sound source at a depth of 91 m along tracks upslope and downslope from the array at ranges from 1–100 km. Slope angles along the tow tracks varied from 0.5°–2.0°. Signal-to-noise ratios of 10 dB and greater were observed on all array elements during much of the source tow. The mode structure of these signals are presented and compared with numerical predictions made by the parabolic equation and fast field program models. [Work supported by NORDA.]

2:35

**II5. Low-frequency sound attenuation in the Mediterranean Sea.** R. H. Mellen (PSI Marine Sciences, New London, CT 06320), T. Akal (SACLANT ASW Research Center, La Spezia, Italy), E. H. Hug (Norwegian Defence Research Establishment, Horten, Norway), and D. G. Browning (Naval Underwater Systems Center, New London, CT 06320)

Sound-channel propagation measurements in the Ionian basin of the Mediterranean Sea have been analyzed to determine the attenuation coefficients in the frequency range of 50–3200 Hz. Concurrent measurements of sound-speed, temperature, salinity, and pH show a strong sound channel having a broad minimum below 100 m and highly uniform properties over the 600-km path. Explosive sources were detonated near the channel axis. The results obtained from hydrophones located near the axis are compared with predictions of the temperature/pH-dependent relaxation-absorption model (components:  $\text{MgSO}_4$ ,  $\text{B(OH)}_3$ , and  $\text{MgCO}_3$ ). Scattering loss is found to be minimal and agreement is good. Attenuation coefficients are compared with earlier values from the Ligurian Sea over a single refraction path approximately 35 km long.

2:50

**II6. Low-frequency propagation across the East Greenland Frontal Zone: Directional dependence of acoustic modes.** Leonard E. Mellberg, Donald N. Connors (Naval Underwater Systems Center, Newport, RI 02841-5047), Ola M. Johannessen (Geophysical Institute, University of Bergen, Norway N-5014), George Botseas, and David Browning (Naval Underwater Systems Center, New London, CT 06320-5594)

In a previous paper [L. E. Mellberg *et al.*, J. Acoust. Soc. Am. Suppl. 1 77, S56 (1985)], an analysis was conducted to determine the relative importance of acoustic propagation modes (bottom bounce, surface duct, and convergence zone) for a shallow source (10 m) along a 100 nmi west to east track through the East Greenland Frontal Zone. In the present paper, a comparison is made to results obtained for a reciprocal east to west track through the zone to determine the directional dependence of the dominant acoustic modes. Along the west to east track, outward from the ice edge and the East Greenland Current, there is a transition from surface duct to convergence zone as the principal propagation mode. This transition, as well as the initial bottom slope, restricts the envelope of ray angles found in this convergence zone mode. For the east to west track, the convergence zone is the dominant mode along the entire path and its character changes significantly. The directional dependence of these modes result in marked changes in propagation loss with range and the energy fields as generated by the two sources. [Work supported by ONR and NUSC.]

3:05

**II7. Determination of sediment shear properties by numerical modeling of ocean-bottom interface waves.** Henrik Schmidt and Finn B. Jensen (SACLANT ASW Research Centre, I-19026 La Spezia, Italy)

The importance of sediment shear properties for low-frequency, shallow-water propagation is well established. Whereas the compressional properties can be measured directly on collected samples, the shear speeds and attenuations have to be determined indirectly from *in situ* experiments. This is due to the fact that these properties are highly affected by the deterioration of the chemical and mechanical bindings due to depressurization and change of temperature in the core sampling process. The

propagation characteristics of ocean-bottom interface waves (Scholte waves) are almost entirely controlled by the shear properties, and these waves, therefore, form a convenient basis for the inversion process. Here, it is demonstrated how the dispersion characteristics, determined from experimental results by a multiple filtering technique, can be numerically modeled by a full wave field solution code [H. Schmidt and F. B. Jensen, J. Acoust. Soc. Am. 77, 813–825 (1985)], yielding the possibility of determining both shear speed and shear attenuation profiles in the upper sediment layers.

3:20

**II8. The determination of geoacoustic models in shallow water.** George V. Frisk, James F. Lynch, and James A. Douth (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A technique for determining the geoacoustic models in shallow water is described. For a horizontally stratified ocean and bottom, the method consists of measuring the magnitude and phase versus range of the pressure field due to a cw point source and numerically Hankel transforming these data to obtain the depth-dependent Green's function versus horizontal wavenumber. In shallow water, the Green's function contains prominent peaks at horizontal wavenumbers corresponding to the eigenvalues for any trapped and virtual modes excited in the waveguide. From the Green's function, one can obtain the geoacoustic model via either forward modeling or perturbative inverse techniques. In the forward modeling approach, a geoacoustic model for the bottom is obtained by computing the theoretical Green's function for various values of the bottom parameters and determining the parameter set which provides the best agreement with the experimental Green's function, particularly in the positions and relative magnitudes of the modal peaks. In the perturbative inverse technique, one uses the differences between the measured modal peaks and those predicted by a background model as input data to an integral equation, which is solved for the bottom geoacoustic parameters. These techniques are demonstrated using experimental data at 140 and 220 Hz. [Work supported by ONR.]

3:35

**II9. The effects of measurement errors in determining the Green's function for shallow water propagation.** James F. Lynch, George V. Frisk, James A. Douth, and Christopher E. Dunn (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

It has been shown by Frisk and Lynch [J. Acoust. Soc. Am. 76, 205–216 (1984)] that, for a horizontally stratified ocean and bottom, one can perform a simple synthetic aperture array cw experiment which yields high-quality information about the modal structure of a shallow water waveguide. Specifically, one measures the magnitude and phase versus range of the pressure field due to a point cw source and then numerically Hankel transforms these data to obtain a depth-dependent Green's function versus horizontal wavenumber. In shallow water, the Green's function contains prominent peaks at horizontal wavenumbers corresponding to the eigenvalues of any trapped and virtual modes excited in the waveguide. In practice, measurement errors can somewhat degrade the quality of the Green's function obtained, as will be shown using both real and synthetic data. Particular errors and experimental effects to be discussed are: ranging errors, nonconstant bathymetry effects, rough surface scattering effects, aliasing and interpolation errors, and the effects of finite aperture. Comparison with conventional array based techniques will also be included. [Work supported by ONR.]

3:50

**II10. Applications of a synthetic aperture array method for shallow-water geobottom reconnaissance.** G. J. Tango, M. F. Werby, and R. Wooten (Naval Ocean Research and Development Activity, Ocean Acoustics and Technology Directorate, NSTL Station, MS 39529-5004)

A general approximate inverse technique for characterizing horizontally stratified shallow-water geobottoms has recently been presented

[Kuperman *et al.*, IEEE J. Ocean. Eng. OE-10(3) (1985); Frisk *et al.*, J. Acoust. Soc. Am. 76 (1984)]. The basic method comprises measuring the pressure field from a cw point source and Hankel-transforming the field numerically to obtain the effective depth-dependent Green's impulse response versus horizontal wavenumber. Because the full wave pressure response  $P$  contains poles at horizontal wavenumbers corresponding to eigenvalues of all discrete, continuous, and evanescent modes,  $G$  can be readily obtained from the empirical transfer function, as derived from a synthetic aperture towed array simulation. Since the inverse Hankel transform simply relates  $P(r)$  and  $G(k)$ , the peaks of the true theoretical Green's function in horizontal wavenumber can be seen to be directly proportional to the discrete angles of maximum beamformer response. Frisk *et al.* have successfully inverted synthetic array Green's function data via least-squares forward modeling of major features via the FFP. Alternatively, an approximate geoacoustic model for the bottom can also be obtained from the modal response angles over near- and farfields. In the specific case of relatively few modes, the highest order of which is near cutoff, locating the discrete/continuous wavenumber demarcation gives the critical angle for the shallowest bottom layer. More generally, selective beamforming over near- and farfields allows modal discrimination from progressively deeper layers for successively increasing source-array offsets. Representative results for accuracy and resolution are given for the cases of single and multilayered isovelocity and gradient-structured geobottoms.

4:05

**II11. Extraction of the shear modulus profile within a sandy seabed by a passive remote bottom shear modulus profiler.** Tokuo Yamamoto (Division of Applied Marine Physics, RSMAS, University of Miami, Miami, FL 33149-1098)

We have developed the software of a bottom shear modulus profiler (BSMP). The BSMP software is a stable and high-resolution inversion procedure for extraction of the bottom shear modulus profile from measured motion of the bed surface induced by propagating gravity waves. Measurements of the motion of a sandy bed and the wave-induced bottom pressure in the water depth of 7.5 m have been made at sea. The shear modulus profile within the sand bed has been extracted from these data using the BSMP inversion procedure. The extracted shear modulus profile agrees well with direct measurements. The paper reports on the experimentation and the inverse analysis of the data. [Work supported by ONR, Code 425UA.]

4:20

**II12. Measurement of complex shear modulus of marine carbonate sand and coral rock by very small amplitude torsional resonant column apparatus.** J. Ludwig Figueroa and Tokuo Yamamoto (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, Miami, FL 33149-1098)

The complex shear modulus of marine carbonate sand and coral rock has been mechanically measured for shear strain amplitudes as low as  $10^{-6}$ . A high precision torsional resonant column apparatus, equipped with the latest electronic instrumentation, was used to determine both the shear modulus and attenuation at various shear strain amplitudes and several stress conditions of prepared marine carbonate sand and coral rock specimens. The influence of test parameters such as confining effective stress and stress anisotropy was studied to simulate actual field states of stress. Vibrational wave amplitude was also varied in order to determine its effect on the dynamic properties of these commonly occurring marine sediments and to compare them with values found in the literature. The frequency range covered was from 50–500 Hz. The geoacoustic data obtained at small shear strain amplitude are of special significance in acoustic wave propagation in the ocean with strong bottom interaction. [Work supported by ONR, Code 425UA.]

4:35

**II13. Measurement of the Biot structural factor  $\delta$  for sintered bronze spheres.** Steven R. Baker<sup>a</sup> and Isadore Rudnick (Department of Physics, UCLA, Los Angeles, CA 90024)

The Biot structural factor  $\delta$  of a porous solid [M. A. Biot, J. Acoust.

Soc. Am. 28, 168 (1956)] can be extracted from measurements of sound speed and attenuation when the solid is saturated with superfluid helium [S. Baker *et al.*, J. Acoust. Soc. Am. Suppl. 1 74, S59 (1983)]. The results of such measurements made in sintered bronze spheres (nom. diam. 75, 110, and 500  $\mu$ ) yield values of  $\delta$  from 3.3 to 4.5. [We take  $\delta$  to be defined by  $\kappa = \delta(\alpha\omega k/\nu P)^{1/2}$ , where  $\kappa$  is the argument of Biot's high-frequency correction function  $F(\kappa)$ ,  $\alpha$  is the tortuosity,  $\omega$  is the angular frequency,  $k$  is the permeability,  $\nu$  is the kinematic viscosity, and  $P$  is the porosity.] If  $\delta_{\text{visc}}$  is the normal fluid viscous penetration depth and  $a$  is a characteristic pore size, then the range of  $(\delta_{\text{visc}}/a)$  covered by these measurements is approximately  $10^{-2}$  to 1. The values of  $\delta$  reported here are about 1½ to 2 times larger than has been estimated for typical marine sediments [R. Stoll, in *Physics of Sound in Marine Sediments* (Plenum, New York, 1974)]. [Work supported by ONR.] <sup>a</sup>Permanent address: Naval Postgraduate School, Code 61Xi, Monterey, CA 93943.

4:50

**II14. Acoustic remote profiling of water interfaces.** J. W. Maxwell and T. L. Henderson (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

Acoustic remote profiling of horizontal surfaces using a monopulse technique has been demonstrated in both a lake and shallow-water ocean environment. Data were taken at a center frequency of 30 kHz using pulses with up to a 5-kHz bandwidth. Excessive variations in the computed profile curves for the lake bottom appeared to be due to penetration of sound into the sediment. Variations in the profile curves decreased when examining an extended planar target deployed on the lake surface. The larger impedance mismatch between the water and air certainly prevented sound penetration beyond the interface. The ocean data profile curves exhibited a bias due to noise effects from surface reverberation. Multiplying averaging reduced profile variability.

5:05

**II15. An improved formula for wavefront curvature ranging based on measuring the normal curvature of wavefronts.** David H. Wood (Naval Underwater Systems Center, New London, CT 06320 and Department of Mathematics, University of Rhode Island, Kingston, RI 02881) and Donald F. Tufts (Department of Electrical Engineering, University of Rhode Island, Kingston, RI 02881)

The conventional formula for wave-front curvature ranging degrades as the source moves away from broadside. This is because it is known [G. C. Carter, J. Acoust. Soc. Am. 65, 528–530 (1979)] that the conventional formula is equivalent to predicting range by triangulation, and this produces poor predictions in the presence of the refraction that is encountered off broadside. An improved formula that partially corrects for refraction is presented. It calculates the so-called *normal* curvature of the wave front—the curvature where the wave front intersects the plane containing both the line through the array of receivers and the line perpendicular to the wave front. This normal radius of curvature is an improved approximation for the range when the source moves away from broadside. The improved and the conventional formulas are compared in examples. [Work supported by the Naval Underwater Systems Center, and the Naval Air Development Command.]

5:20

**II16. Function theoretic methods for acoustic wave propagation in the ocean.** Robert P. Gilbert (Department of Mathematical Sciences, University of Delaware, Newark, DE 14716 and Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320) and David H. Woods (Department of Mathematics, University of Rhode Island, Kingston, RI 02881)

In this approach to modeling the sound propagation in the ocean, *function theoretic methods* are used in partnership with *numerical methods*. The problem of computing a sound field in the ocean is greatly simplified by using a two-pronged approach involving (1) function theoretic methods that analytically incorporate the major features of the ocean and its bounding surfaces, including the radiation condition and the average index of refraction, and (2) numerical methods to model the effects of the

secondary but detailed features of the ocean and its boundaries. For example, to model a sound field in a uniform ocean with perfectly reflecting boundaries over a long distance is a difficult numerical problem but a trivial analytical problem. In our two-pronged approach, numerical tech-

niques are used only to compute the *changes* that are observed as a result of perturbing the ocean from its analytical idealization. [This work was supported by Code 10 and Code 3332 of the Naval Underwater Systems Center.]

THURSDAY AFTERNOON, 7 NOVEMBER 1985

DAVIDSON ROOM A, 2:00 TO 4:25 P.M.

### Session JJ. Engineering Acoustics III: Modeling of Large Arrays

Stephen C. Thompson, Chairman

Gould Defense Systems, Inc., 18901 Euclid Avenue, Cleveland, Ohio 44117

Chairman's Introduction—2:00

#### Invited Papers

2:10

**JJ1. Analyses of large arrays: Brief theory and some techniques used in 1954–1985.** Gordon E. Martin (Martin Acoustics Software Technology, P. O. Box 86050, San Diego, CA 92138-6050)

The behavior of underwater acoustic arrays of sources and receivers is reviewed with the historical perspective of some applications during the last 30 years. The general basis for interelement interactions of acoustic, mechanical, and electrical components is described. Array analyses require the solution of large matrix equations. This is done with large computer assets (not available 20–30 years ago), or special symmetry and special algorithms. The mathematical matrix form for several types of array systems is shown including particularly those with Toeplitz symmetry. The author's early (1954) formulation of an array problem with very adverse interactions including elements with negative radiation resistances is given as a simple illustration for a small array. Modular and nonmodular electrical drives have different matrix forms. The techniques for computation of the performance of arrays with many elements including the Trench algorithm (1960's) are described. See abstracts of the author's papers given at prior meetings. Results for large arrays include damage due to adverse interactions, discovery and development of velocity control techniques, and difficulties due to electrical steering of directional response. Slides include the 1955 nearfield calibration of a planar source array with 200 elements. Some results are given for an array of more than 1000 elements, having block Toeplitz form with Toeplitz blocks.

2:40

**JJ2. Historical account of mutual acoustic radiation interaction between elementary transducers of a large underwater sound propagator array.** S. Hanish (Code 5104, U. S. Naval Research Laboratory, Washington, DC 20375)

In the decade 1955–1965 the U. S. Navy through ONR funded the construction of the ARTEMIS projector array for conducting long-range, low-frequency propagation experiments in the ocean. The acoustic size ( $= ka = 2\pi a\lambda$ ) of the elementary transducers was very much less than unity at the transmission frequency. Early trials revealed extensive difficulty in achieving high-power radiation with uniform electrical excitation. It was soon found that the acoustic load varied significantly from element to element, leading to element failure, low transmission efficiency, and operating difficulty with the power amplifier. A historical review is presented for the valuable lessons learned in this experience.

3:10

**JJ3. Effects of nonrigid mounting surfaces on array behavior.** David T. Porter (Naval Underwater Systems Center, New London, CT 06320)

Three problems are considered. (1) The general problem, whereby the transducer array excites the mounting structure, which reradiates, and the resulting acoustic field is diffracted by the transducers. (2) The simplified problem whereby the transducers are much smaller than a wavelength, and scattering off of the transducers can be neglected. (3) The simplified problem whereby the transducers are acoustically connected by mutual coupling through their radiating faces, and mechanically coupled through their tail ports and the mounting surface, but acoustic waves do not strike the mounting surface. For a certain class of mounting surfaces, the third problem can be carried out. Calculations were performed when the mounting structure was an infinite elastic flat plate, or a simply supported beam. A hypothetical line array of eight elements was considered, driven at 9 kHz; the thickness of the mounting surface was varied, to alter the stiffness of the surface. Large variations in array behavior were seen for thicknesses between 0.5 and 5.0 in.

3:40

**JJ4. Mutual radiation impedance of multiple mode vibrators.** George W. Benthien (Naval Ocean Systems Center, Code 712, San Diego, CA 92152)

Most analyses of mutual radiation coupling to-date have assumed that the velocity distribution on each piston is fixed and known. In some problems, however, the velocity distribution on each radiator is made up of several modes and the modes interact acoustically as well as mechanically. For example, the velocity distribution on each piston might consist of a rigid body mode and one or more flexural modes. In this paper a general framework will be presented for the modal acoustic interaction problem, and several numerical techniques for its solution will be examined.

3:55

**JJ5. Mutual radiation impedance of baffled pistons.** George W. Benthien (Naval Ocean Systems Center, Code 712, San Diego, CA 92152)

The mutual radiation impedance of pistons on a plane infinite rigid baffle is expressed as a Fourier transform of an impulse response. The technique used is an extension of the work of Lindemann on the self-radiation impedance of baffled pistons [O. A. Lindemann, *J. Acoust. Soc. Am.* **55**, 708-717 (1974)]. As an illustration, the results are applied to the mutual interaction of rectangular pistons on a plane. Numerical results

are presented and compared with other published results in order to demonstrate the accuracy of this technique.

4:10

**JJ6. Finite element study of the farfield limit of symmetrical length expander transducer.** B. Tocquet, D. Boucher (GERDSM, 83140 Le Brusc, France), P. Tierce, and J. N. Decarpigny (ISEN, 3 rue Francois Baës, 59046 Lille Cedex, France)

The finite element code ATILA which has been described previously [*J. Acoust. Soc. Am. Suppl.* **1** **74**, S99 (1983)] and is devoted to the modeling of radiating sonar transducers takes account of the acoustic radiation with the help of a finite element discretization of the fluid domain which surrounds the structure. The fluid domain is limited to a spherical exterior boundary upon which a suitable nonreflexive condition is applied. Due to its simplicity, the nonreflexive condition of a spherical wave is generally used and thus the exterior boundary has to be in the farfield region. This communication describes the numerical determination of the farfield limit in the case of a thick transducer which is composed of a long ceramic stack between two identical head masses and has a classical length expander fundamental mode. Due to the thickness of the source, the classical Rayleigh criterion is incorrect and the farfield limit depends upon the type of housing which is used. The various results are described and discussed, from the physical point of view as well as from its numerical consequence.

THURSDAY AFTERNOON, 7 NOVEMBER 1985

SUITE 4, 2:00 TO 5:00 P.M.

## Session KK. Musical Acoustics III

Uwe J. Hansen, Chairman

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

## Contributed Papers

2:00

**KK1. A critical look at the coupling between the so-called "Helmholtz air" mode and a beam mode of the whole violin.** Carleen M. Hutchins (Catgut Acoustical Society, Inc., 112 Essex Avenue, Montclair, NJ 07042)

A study of response curves, made over the last 20 years, of violins of varying musical qualities shows that in most of the instruments owned and considered to be of excellent quality by professional players, there is a close frequency relationship between the "Helmholtz air" mode and that of the beam mode around 270-290 Hz. When these two modes occur at the same frequency, there is enhanced amplitude of the "Helmholtz" resonance as well as increased vibration of the whole instrument, not only at this frequency but also at higher frequencies, as shown by response curves. In addition, this close coupling affects the "feel" of the instrument in the player's hands, particularly through the enhanced vibration of the neck. A quick way to check the relation of these two modes is to hold the violin upside down in thumb and forefinger at about half the distance between the bridge and saddle and alternately tap on the end of the scroll and blow in one f hole. With some practice the pitch of these two modes can be compared and also heard with considerable clarity when their frequencies coincide. Implications for violin making and possible effects on bowing will be discussed and a violin demonstrated.

2:15

**KK2. A dynamic analysis of the violin using finite elements.** George Knott (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

The dynamic response of selected points on a 12 000 degree-of-freedom finite element model of a violin to a simulated bowing force was studied. The model consists of the body (plates, bassbar, soundpost, end and corner blocks, ribs, and rib linings), neck, bridge, tail piece, and strings. All components were modeled as elastic elements *in vacuo*. Comparisons between the frequency response of a real violin and the model will be discussed.

2:30

**KK3. Timpani normal modes for arbitrary shaped kettles.** Robert E. Davis and Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

Calculations of timpani normal modes based on boundary integral equation methods [R. S. Christian, R. E. Davis, and A. Tubis, *J. Acoust.*

Soc. Am. Suppl. 1 73, S85 (1983)] are carried out for a variety of cylindrically symmetric kettle shapes. The kettle is assumed to be rigid, and room acoustics effects are ignored. The kettle shapes considered include hemispherical, hemispherical cap on a cylindrical sleeve, and parabolic. We extend previous studies [R. S. Christian *et al.*, J. Acoust. Soc. Am. 76, 1336-1345 (1984)] of (1) the approximate equivalence of air loading effects for equal membrane radii and kettles of equal volumes; and (2) the minimal kettle volume for producing the musically desirable ratios,  $f_{11}:f_{21}:f_{31}:f_{41} \approx 2:3:4:5$ .

2:45

**KK4. Group classification of vibration modes in bells.** Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115) and Robert Perrin (Department of Physics, University of Technology, Loughborough, LE11 3TU, England)

Several different systems have been used in classifying the vibration modes of church bells, carillon bells, and handbells. No existing classification system, however, has been completely satisfactory in describing modes of vibration in all types of bells. In this paper we describe a new mode classification system which is applicable to small handbells as well as to large church bells and carillon bells. The well-known inextensional modes are arranged into groups (0, I, II, ...) according to the number and location of their nodal circles, and a "periodic table" is constructed. Less familiar torsional, breathing, swinging, extensional, and ring axial modes are arranged in groups having similar modal shapes.

3:00

**KK5. Tang influence on handbell mode.** Uwe J. Hansen (Department of Physics, Indiana State University, Terre Haute, IN 47809), Scott Hampton, and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Manufacturers routinely cast handbells with a stem at the crown, identified as the tang, facilitating the machining and tuning of the bell. For some models, portions of the tang are retained for handle and clapper fastening, while in others, attachments are made through a hole in the crown. The effect of the tang on the frequencies and amplitudes of musically significant modes is of interest to both handbell ringers and to scientists concerned with an understanding of the tonal structure of handbells. Modal mapping, using nearfield sound radiation and time-averaging holographic interferometry, was used to study G4, F6, and D7 handbells with and without tang. Tang removal lowered the frequency of the fundamental by approximately 1.3%. Holographically observed mode shapes for the musically significant, lowest modes (2,0; 3,0; and 3,1) indicate very little vibrational encroachment into the crown area, suggesting that the crown and with it the tang, do not play a significant role in handbell timbre.

3:15

**KK6. Digital music synthesizers as perceptually valid phenomenological models.** G. L. Gibian, M. Ben Mohamed, and E. N. Harnden (Physics Department, American University, Washington, DC 20016)

Commercially available digital music synthesizers can produce sounds which perceptually resemble many traditional acoustic instruments. As with all phenomenological modeling, the quality of the output (of a synthesizer, in this case) does not require the generation process to incorporate the physics of the modeled systems (which are acoustic musical instruments in this case). Spectrum-versus-time and spectrum-versus-pitch plots for several commercially manufactured digital synthesizers employing different synthesis techniques (including audio-rate frequency modulation) will be compared with their counterparts from traditional acoustic instruments. It is also of interest to compare the spectra after imposing transformations which model psychoacoustic processing, such as loudness curves and integration within critical bands.

3:30

**KK7. Scheduled language for synclavier.** W. A. Burnette and M. V. Mathews (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A real-time music performance language has been written for the New

England Digital Synclavier II. It is based on RTSKED, a scheduled language previously implemented on a Z80 microprocessor. The language allows easy and precise control of the timing of computer controlled events. Events can be controlled by external inputs such as pressing a key, or by an internal clock. Several sequences of events can operate simultaneously to generate complex musical sounds. A schedule is a "pure procedure" that consists of lists of operations, including transfers and conditional transfers. An "active schedule" comprises a set of variables describing the computer's progress through the execution of a schedule. An active schedule may create other active schedules and may generate or wait for "triggers," which are notifications of events from external inputs, internal timeouts, or other active schedules. Sounds may be generated by the combined efforts of several active schedules using schedule parameters and real time sensor inputs to compute a sequence of output channel commands. The language was demonstrated by performing four compositions written by R. Rocco.

3:45

**KK8. Impulse-response measurements of small acoustical systems using shift-register sequences.** Rodney D. Price and E. Paul Palmer (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

A data-processing system, using an IBM PC/XT, was developed to obtain impulse-response measurements of small acoustical systems such as musical instruments or the human vocal tract. The test signal is an  $m$  sequence generated using a shift register and converted to an acoustic plane wave by a piezoelectric transducer/tube system. We compensate for the nonideal characteristics of the transducer and microphone by using inverse filters generated with an adaptive digital filter [B. Widrow *et al.*, Proc. IEEE 63, 1692-1716 (1975)]. The impulse response is recovered with a correlation technique using Hadamard matrices [M. Cohn and A. Lempel, IEEE Trans. Inf. Theory 23, 135-137 (1977)]. We present impulse-response measurements, 8191 samples in length at 24-kHz sampling rate, for various tube systems. Extension of the technique to reverberant systems with long impulse responses is straightforward since the data are recorded directly on disk to avoid memory-length limitations. The correlation algorithm, measurement results, and limitations of the transducer-compensation technique are discussed.

4:00

**KK9. A new method for the measurement of acoustic impedance and its application to musical instrument research.** Anne-Marie Bruneau (Laboratoire D'acoustique, Faculte des Sciences, University du Maine, 72017 LeMans, France) and George R. Plitnik (Frostburg State College, Frostburg, MD 21532)

A method to accurately measure acoustic impedance over an extended frequency range without the limitations imposed by capillary tubes is described. The apparatus consists of two electrostatic microphones, one used as a source of excitation and one as receptor. By means of the equations of reciprocity, the frequency dependence of an unknown acoustic impedance may be accurately deduced. Experimental results and theoretical calculations will be compared for several well-known acoustic transmission line systems, and the results of utilizing this system to measure the input impedance of a musical horn will be presented. The advantages of this system over capillary tube measurements for musical instrument research, will be discussed, and future applications will be mentioned.

4:15

**KK10. Spectral envelopes of orchestral instruments.** A. H. Benade and S. N. Kouzoupis (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

Dependable techniques are available for the measurement of musical spectra. The oscillation and radiation dynamics of wind, bow, hammer, and plectrum excited instruments is well understood. Most of an instrument's dynamical system is shared by the production processes of all its notes, so that similar spectrum envelopes are expected for many notes in its scale. Measured spectra confirm the expectation, and show that these similarities are also shared by most instruments of the same playing range.

Thus the notes of violin, piano, trumpet, oboe, and clarinet (odd partials) scales have envelopes caricatured by  $E_0(f) = [1 + (f/f_c)^2]^{-1/2}$ , with  $\beta = 3$  and  $f_c = 1500$  Hz. Corresponding alto instruments, e.g., English horn, alto sax, and alto clarinet, tend to have  $f_c$  close to 1000 Hz, and  $\beta = 3$ . Several mechanisms, each with its own slope and breakpoint, combine in each instrument to produce its  $E(f)$ , yet makers and players normally choose combinations that approximate  $E_0(f)$ . This suggests perceptual constraints on desirable tone color, e.g., via the "sharpness" integral of  $v$ . Bismarck, refined by Terhardt *et al.* Numerous measured envelopes will be presented, and discussed via their sharpness functions for treble, soprano, alto, and bass instruments. [Work supported by NSF.]

4:30

**KK11. The influence of room transients and traversal on loudness discriminability in rooms.** Ian M. Lindevald and A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

Six variants of the generic loudness discrimination experiment were conducted to study the influences of room transients and subject mobility in discrimination in rooms. Subjects judged the louder of a pair of tones as the source level difference  $\Delta$  was varied. The fraction of wrong judgments was plotted against  $\Delta$ , and the rms spread  $\sigma$ , and the total percent wrong judgments  $W$  were found. The six variants were as follows: (1) stimuli presented via two loudspeakers to subjects moving in a room,  $\sigma = 1.87$  dB,  $W = 10\%$ ; (2) stimuli produced in room and recorded via moving dummy head, presented via headphones,  $\sigma = 1.98$  dB,  $W = 11\%$ ; (3) gated signals channeled from oscillator to earphones,  $\sigma = 0.50$  dB,  $W = 6\%$ ; (4) steady tones played in room, recorded via moving dummy head, then gated,  $\sigma = 1.52$  dB,  $W = 8\%$ ; (5) gated signals played in room recorded

via stationary dummy head,  $\sigma = 2.32$  dB,  $W = 12\%$ ; and (6) samples of steady tones recorded with a stationary dummy head at different points in the room and then gated,  $\sigma = 2.72$  dB,  $W = 33\%$ . The auditory system can discriminate source strength in rooms via data from the on/off transients and also from relatively long term sampling from different locations. Removal of such clues reduces discriminability to levels controlled by ordinary room statistics. [Work supported by NSF.]

4:45

**KK12. Absolute tonality versus absolute piano.** W. Dixon Ward (Hearing Research Laboratory, University of Minnesota, 2630 University Avenue SE, Minneapolis, MN 55414)

Earlier studies of the ability of individuals to recognize when excerpts from Bach's "Well-Tempered Clavichord" has been shifted upward or downward from the original key by 1 or 4 semits suggested that almost all musicians were able to recognize the direction of these shifts at a better than chance level, even those who professed little or no ability to identify isolated tones (absolute pitch). In an effort to determine to what extent this ability represents (1) "absolute tonality"—long-term memory for the specific compositions, (2) detection of a disparity between sound and printed score, and (3) a vague recognition that the series of tones, no matter who wrote them, are played too high or too low to be good music, ten musicians were given the test "cold"—i.e., they were asked to identify the shifted excerpts with no prior exposure to examples of extreme shifts and without being shown the score. All succeeded in identifying the  $\pm 4$ -semit transpositions, though not the  $\pm 1$ -semit ones. A week later, a second test was given, but with the printed score (in the original key) available. Now, even the  $\pm 1$ -semit shifts were recognized. The results provide little support for the absolute tonality hypothesis.

THURSDAY AFTERNOON, 7 NOVEMBER 1985

DAVIDSON ROOM B, 2:00 TO 5:00 P.M.

## Session LL. Physical Acoustics VII: Scattering

Roger H. Hackman, Chairman  
Naval Coastal Systems Center, Panama City, Florida 32407

### Contributed Papers

2:00

**LL1. Acoustic scattering in an ocean environment.** Roger H. Hackman and Gary S. Sammelmann (Naval Coastal Systems Center, Panama City, FL 32407)

The transition matrix approach has been extensively applied to the free-field acoustic scattering from elastic solids and shells in the past decade. However, it is not known to what extent the salient features of the free-field scattering problem are altered when the scattering problem is reformulated in a realistic shallow water or ocean environment. The present work is centered on extending the transition matrix approach to the case of acoustic scattering by an elastic obstacle immersed in a stratified ocean environment. The ocean is assumed to be vertically bounded and propagation aspects are treated with a normal mode model. Numerical studies are presented for the acoustic scattering from a solid elastic target in the two simplest oceanic waveguide models of interest. The first is a single homogeneous layer bounded by a pressure-release air-water interface and a rigid bottom. The second is a double-layered waveguide to

model a single-channel sound-speed profile with a depth excess. [Work supported by the Office of Naval Research.]

2:15

**LL2. Eigenfrequencies of elastic cylinders and shells; surface waves generated by normally and non-normally incident sound.** J. V. Subrahmanyam,<sup>4)</sup> H. Überall, and A. Nagl (Department of Physics, Catholic University, Washington, DC 20064)

We obtain the characteristic equation of an elastic cylinder, or cylindrical shell, immersed in a fluid and enclosing another fluid for vibrations containing both azimuthal and axial components. Numerical solutions are found for aluminum and steel cylinders, in the form of distinct "layers" of roots in the complex frequency plane, which may be interpreted as the resonances of azimuthal or helical surface waves around the shell. Rayleigh waves, Stoneley waves, and Lamb (whispering gallery) waves are



identified and the interaction of Rayleigh and Stoneley branches at low frequency is clarified. Surface wave speeds and attenuations are obtained as functions of frequency. [Work supported by the Office of Naval Research.]<sup>1</sup> J. V. Subrahmanyam is now at Osmania University, Hyderabad 500 007, India.

2:30

**LL3. Experimental study of interface waves on a curved surface.** F. Luppé, G. Quentin (G.P.S., Université Paris 7, Tour 23, 2 place Jussieu, 75251 Paris Cedex 05, France), A. Stoyanov (Physics Department, Catholic University, Washington, DC 20064), and H. Überall (Physics Department, Catholic University, Washington, DC 20064 and G.P.S., Université Paris 7, Tour 23, 2 place Jussieu, 75251 Paris Cedex 05, France)

We present an experimental study of surface waves generated on a curve surface at grazing incidence. We have observed three types of circumferential waves travelling at a velocity close to that of water on the external surface of an elastic cylinder imbedded in water. Attention is focused on the attenuation and reemission of these waves in the surrounding fluid. We then study the influence of a discontinuity of the curvature on the propagation of these waves. [Work supported by the Direction des Recherches, Etudes et Techniques, France, and by the U.S. Office of Naval Research.]

2:45

**LL4. A study of angular distributions from submerged spheroidal and finite cylindrical shells.** M. F. Werby (Naval Ocean Research and Development Activity, NSTL, MS 39529), L. H. Green (Martin Marietta Corporation, 103 Chesapeake Park Plaza, Baltimore, MD 21220), and R. P. Wooten (ODSI, Rockville, MD 20852 and NORDA, NSTL Station, MS 39529)

In the investigation of submerged objects, one generally examines backscattered form functions. In theoretical studies, this has been standard practice in the investigations of spheres and infinite cylinders; and this usually limits one to the determination of resonances. When such studies are carried out, one must already know the orientation of the incident field relative to the object (this is obvious for spheres); and calculations must be performed over a broad frequency range in small increments. However, for elongated objects, the orientation of the incident field relative to the object as well as the actual shape of the object is of importance and cannot be determined from backscattered measurements alone. We examine the feasibility of determining shape information as well as orientation by studying bistatic angular distributions from spheroids and finite cylinders for a variety of orientations relative to the axis of symmetry. In addition, we examine the effect resonances have on angular distributions, particularly in the forward direction.

3:00

**LL5. Scattering from submerged solid-, fluid-, and air-filled elastic shells to the high  $ka$  region.** G. Gaunard (NSWC, R-43, White Oak, Silver Spring, MD 20910), C. Feuillade (ODSI Defense Systems Inc., 6110 Executive Boulevard, Rockville, MD 20852), and M. F. Werby (Naval Ocean Research and Development Activity, Code 220, NSTL, MS 39529)

We examine scattering from submerged, solid-, fluid-, and air-filled elastic spheres for  $ka$  values ranging from 0 to 100 (where  $k$  is the wave-number of the incident waves and  $a$  is the radius of the sphere). The analysis includes interpretation of forward and backscattered form functions for a variety of materials and shell thicknesses. An analysis of resonances, particularly their change in character with increasing  $ka$ , will be illustrated using numerous examples.

3:15

**LL6. The interaction of acoustic waves with multi-layered shells.** L. H. Green (Martin Marietta Aerospace, Baltimore Division MP405, 103 Chesapeake Park Plaza, Baltimore, MD 21220), M. F. Werby, and R.

P. Wooten (Naval Ocean Research and Development Activity, NSTL, Bay St. Louis, MS 39529)

We report on our investigation of steady-state scattering of acoustic waves by multi-layered shells with an elastic inclusion. Numerical results will be presented for scattering from spheroidal shells and finite cylindrical shells with hemispherical endcaps. Comparisons will be made with similar results for scattering from elastic solids, thin shells, and impenetrable targets. The frequency of the incident acoustic wave will be in the resonance region of the materials studied. The identification of prominent effects with the elastic waves propagating in the shell will be made. The dependence of the results on frequency, shell and fluid parameters, and aspect ratio is discussed. A physical interpretation of the results will be provided.

3:30

**LL7. Analysis of the impedance boundary condition in terms of an exact scattering calculation.** Virginia M. Ayres and Guillermo C. Gaunard (Naval Surface Weapons Center, R43, 10901 New Hampshire Avenue, Silver Spring, MD 20903-5000)

The scattering of incident acoustic waves by convex fluid-loaded bodies can be analytically modeled at various levels of realism. In our approach to scattering from elastic/viscoelastic bodies, the coefficients of the exact Rayleigh series for the scattered field are determined by the application of three boundary conditions (b.c.). These boundary conditions are statements of continuity of stresses and displacements across the body's surface. An often-used approximation assumes the body to be impenetrable and subjected to a single impedance-type or Cauchy b.c. on its convex surface. The specific surface impedance of the body's material, usually determined by other means, is introduced into the formulation in an "ad hoc" way. The details and limitations of how a single simple b.c., on a body assumed impenetrable, can model the effects of three quite complex ones, on a realistically modeled penetrable object, are studied here for a simple (spherical) shape. Comparison of the two methods yields an expression for the surface impedance which is dependent on frequency, on mode order, and on other parameters of the acousto-elastic solution. We find this dependence to be quite complicated. The outcome, which is amply supported by numerical displays, provides a more basic understanding of the uses and limitations of the impedance approach to predict echoes from any fluid-loaded real object.

3:45

**LL8. Bistatic echoes from elastic bodies in the resonance scattering regime.** C. Y. Tsui, G. N. Reid, and G. C. Gaunard (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20903-5000)

The experimental technique that we described earlier [C. Y. Tsui *et al.*, J. Acoust. Soc. Am. Suppl. 1 77, S79 (1985)] which extracts the active resonance spectrum of any scatterer from the echo it returns, was applied to solid elastic cylinders and to cylindrical shells of various thicknesses immersed in water. This technique consists in sampling the tailend of the returned pings, which contain the elastic (whispering gallery) resonances of the bodies, in order to isolate them from the (rigid) backgrounds they are usually mixed with. We now study various experimentally obtained differential scattering (bistatic) cross sections for the above mentioned objects. It is found that at certain resonance frequencies, the background-suppressed, bistatic cross section, which is a measure of the free-vibration target reradiation, ideally consists of a symmetric rosetta pattern, having twice the number of lobes as the modal order  $n$  of the body resonance being excited. Using theoretically computed plots of the actual spectral shapes of the body's individual modal resonances, we explain that the anomalous angular plots, which have more than  $2n$  lobes, are caused by a superposition of several modes for which other broad modal resonances having wide tails, interfere with the chosen resonance frequency within the  $n$ th mode. Theory and experiment complement each other to elucidate the resulting aliasing effect.

4:00

**LL9. Resonance spectra of elongated elastic objects.** H. Überall, A. Nagl, Y. J. Stoyanov (Department of Physics, Catholic University,

Washington, DC 20064), S. K. Numrich (Naval Research Laboratory, Washington, DC 20375), S. H. Brown, J. W. Dickey (David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402), and J. M. D'Archangelo (Department of Mathematics, U.S. Naval Academy, Annapolis, MD 21402)

It is well known [H. Überall, L. R. Dragonette, and L. Flax, *J. Acoust. Soc. Am.* **61**, 711 (1977)] that the complex eigenfrequencies at which smooth convex objects resonate under the incidence of an acoustic wave are those at which circumferential waves generated by the incident signal phase match over a closed orbit. This principle was verified by us, by analytically obtaining the resonance frequencies of elastic cylinders and spheres [E. D. Breitenbach *et al.*, *J. Acoust. Soc. Am.* **74**, 1267 (1983)] and deriving phase and group velocities of the surface waves from this. In the present study, this approach is inverted, and applied to elastic prolate spheroids and to cylinders with hemispherical endcaps. From the known phase velocities and trajectories (geodesics) of the surface waves, we were able to predict the elastic resonance frequencies of these bodies. [Work supported by the David W. Taylor Naval Ship R & D Center, the Naval Research Laboratory, and the Office of Naval Research.]

4:15

**LL10. Resonances of a spherical shell covered with a constrained viscoelastic layer.** D. E. Husson (G.E.R.D.S.M., Le Brusc, 83140 Six Fours les Plages, France)

A computer model has been written to predict the interaction of a plane acoustic wave with an elastic or viscoelastic layered sphere. This model is based on the decomposition of the acoustic potentials in each layer in spherical harmonics. It can compute the scattered near- and far-fields of spheres formed of up to ten layers, at frequencies up to  $ka = 120$ ,  $a$  being the radius of the sphere and  $k$  the wavenumber of the incident wave. This model is used to study the influence of a constrained layer, consisting of a viscoelastic layer and a constraining outer shell, on the resonances of a metallic hollow sphere. Viscoelastic materials are modeled according to the equations of the Kelvin-Voigt model, with complex elastic moduli. Plots of the acoustic pressure backscattered by the sphere, of the total pressure on its surface, of its surface displacements, and color-coded maps of the pressure around the sphere are presented to show the influence of the thickness of the viscoelastic layer and of its absorption coefficients on the resonances of the metallic shell, and the effect of the resonances of the three layers on the scattered field.

4:30

**LL11. Scattering from multiple compliant tube gratings in an elastic layer.** Ronald P. Radlinski and Robert S. Janus (Naval Underwater Systems Center, New London, CT 06320-5594)

From previous studies of scattering from multiple gratings of resonant compliant tubes in water [R. P. Radlinski and M. M. Simon, *J. Acoust. Soc. Am.* **72**, 607-614 (1982)], the excitation of noncompliant, antisymmetric structural modes by nearfield evanescent waves was found to severely degrade the reflectivity of closely packed gratings in the bandwidth of excitation of the compliant symmetric modes. Also, transmission resonances due to the spring-mass-spring configuration of the two gratings separated by a fluid mass diminished low-frequency performance. In this paper, encapsulating the gratings in a low-stiffness elastomer is shown experimentally to have a minimum effect on single and widely separated gratings with respect to fluid but enhances the performance of closely packed gratings. Comparison of insertion loss performance with a high stiffness encapsulant indicates dramatic frequency response differences for closely packed arrays. Two different grating configurations will be considered and comparisons with a mathematical model will be discussed.

4:45

**LL12. Reflection and transmission characteristics of a periodic array of cylinders embedded in a plane slab or in a homogeneous material space.** Akhlesh Lakhtakia, Vasundara V. Varadan, and Vijay K. Varadan (Laboratory for Acoustic and Electromagnetic Research, Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

A periodic array of identical cylinders, identically oriented, and embedded either inside a plane, infinite slab suspended between two homogeneous half-spaces, or by themselves embedded in a material space, forms an interesting class of gratings whose periodicity is vertically pronounced. A theory based on Fourier-Bessel expansions and the  $T$ -matrix method is presented to compute the reflection and transmission characteristics of such a grating. The theory is sufficiently general so as to be applicable to electromagnetic incidence of either polarization or to elastic incidence having the  $SH$  polarization. The cylinders may be isotropic or may possess transverse isotropy only. Numerical results presented will show that such a grating reflects and transmits specularly almost perfectly, which is in contrast with more conventional gratings made up of periodic, bimaterial interfaces and whose periodicities are horizontally pronounced.

**Session MM. Engineering Acoustics IV: Acoustic Signal Processing Implementations**

James F. Bartram, Chairman

*Raytheon Corporation, Submarine Signal Division, P.O. Box 360, Portsmouth, Rhode Island 02871***Chairman's Introduction—8:30***Invited Papers***8:40**

**MM1. A real-time digital signal processing evaluation system.** W. Vance McCollough (Hughes Aircraft Co., 8000 E. Mapewood Avenue, Englewood, CO 80111), Diane M. Knight, David Erickson, and Raymond A. Jannsen (Raytheon Co., SSD, P.O. Box 360, Portsmouth, RI 02871)

This presentation will describe a test-bed processing system which is based on four serially connected Texas Instruments TMS-320 signal processors. Programs can be developed using standard TI development tools, and loaded into the test-bed processor for execution. Any one of the processors can be replaced by the TI EVM in-circuit emulator, for more flexibility in algorithm testing and development. A variety of status, control, and interrupt inputs and outputs are provided to interface the processors to each other and to external devices. Built-in A/D and D/A converters provide ports for data acquisition and output, and can be synchronously clocked from external sources. Digital data can also be input and output—the digital output port can directly interface to a line scan recorder. The primary uses for the test-bed processing system are to test software and algorithms on real data and in a realistic hardware environment, and as a laboratory tool to analyze experimental data. An example will be discussed which shows how the test-bed processor has been used in the above ways to evaluate a correlation processor, and to process experimental data.

**9:10**

**MM2. Hardware implementations of real-time digital speech processing algorithms.** Elliot Singer (Lincoln Laboratory, Massachusetts Institute of Technology, Lexington, MA 02173-0073)

Continuing advances in hardware technologies are permitting the realization of increasingly sophisticated speech processing algorithms in real-time equipments. The availability of commercial digital signal processing integrated circuit components has been especially responsible for a reduction in the size and cost of these devices. This presentation will describe the unique requirements of speech compression and speech recognition algorithms with respect to arithmetic calculations, memory, and I/O. Representative equipment designs developed at Lincoln Laboratory for realizing real-time speech processing algorithms will be described. These include: the Lincoln digital signal processor (LDSP), a programmable, general-purpose ECL machine suited for real-time evaluation of speech processing algorithms; the advanced linear predictive coding microprocessor (ALPCM), a flexible bit-slice processor designed for use in operational environments; and the compact linear predictive coder, a small, narrow-band vocoder terminal based on DSP microprocessors. The application of advanced VLSI technology to meet the processing demands of large vocabulary speech recognition will be discussed, with specific focus on an approach being pursued at Lincoln Laboratory which uses wafer scale integration and restructurable VLSI technology to exploit the high level of concurrency in the recognition algorithm. [Work sponsored by the Department of the Air Force.]

**9:40**

**MM3. Hardware for time delay beamforming, demodulation, and processing.** K. Metzger, Jr. and T. G. Birdsall (Communications and Signal Processing Laboratory, The University of Michigan, 2355 Bonisteel Boulevard, Ann Arbor, MI 48109)

A digital time delay beamformer/demodulator/processor was designed and built as part of the U of M's involvement in ocean acoustic tomography measurements. This paper describes the overall system design, the criteria on which design decisions were made, and how well the resulting equipment performs. The beamformer can form up to eight simultaneous beams using up to 24 hydrophone inputs per beam. The beam outputs are coherently shifted from any center frequency in the range from 10 Hz to over 500 Hz down to baseband. Center frequencies can be set in steps of 1 mHz. Special hardware was included in the system to speed up commonly performed processing operations. The fast arithmetic device uses serial multipliers and adders to aid in performing double precision complex additions and various multiplication operations. The sequence removal hardware speeds up the computations involved in pulse compressing the linear maximal sequence coded transmissions commonly used in making ocean acoustic multipath measurements. The speed-up factor is about 100. [Work supported by ONR.]

**MM4. BBN Butterfly applications to signal processing.** Francis J. M. Sullivan (BBN Laboratories Inc., San Diego, CA 92110), H. Briscoe, R. Estrada, and E. Schmidt (BBN Laboratories Inc., Cambridge, MA 02138)

Modern digital signal processing systems require substantial hardware and software flexibility at a sufficiently low cost. The cost of configuring the hardware and developing the software should be low, and it should be easy to make revisions as requirements such as the number of channels and data rates change. Also important is efficiency for both numerically intensive operations, such as FFT and filter operations, and logical and symbolic manipulation intensive operations such as decision making and display generation. This mixture of computationally intensive operations is ideally suited to concurrent parallel processing. The BBN Butterfly multiprocessor provides a general purpose parallel processor which may be configured for a wide range of applications. It contains up to 256 simultaneously operating asynchronous processor nodes, each containing a general purpose microprocessor, local memory, and optional floating-point accelerator, optional high-speed auxiliary processors, and a unique expandable inter-processor communication system. The tight processor coupling and global memory sharing provided by the communication system allows the necessary coordination of the operations in the independent processor nodes, and provides a software environment which dramatically facilitates the ease of programming.

### Contributed Papers

10:40

**MM5. Measurement of aliasing in cepstrum analysis.** Mahmoud N. Fahmy (College of Computer and Information Sciences, King Saud University, Riyadh-11543, Saudi Arabia)

In acoustical signal processing, cepstrum analysis is used to deconvolve an output signal into a path impulse response and an input source wavelet. The cepstrum of an input wavelet will occupy the first portion in the queffrequency domain while the path response will occupy the last portion. Aliasing in the queffrequency domain is of course an ever-present problem. This aliasing is caused by introducing a nonlinear complex logarithm in the frequency domain. The aliasing error has a significant effect on the last portion in the queffrequency domain, which represents the path response. In this work, a special proposition is introduced to measure the error due to aliasing caused by sampling of nonband-limited function in the queffrequency domain. By appending zeroes to the frequency domain or multiplying the time domain by an exponential window, the aliasing error in a path response can be reduced. Different examples using a real signal will be presented. A special type of long pass lifter was used to extract the path response with a minimum of aliasing errors measured by the new proposition.

10:55

**MM6. Statistical signal analysis for systems with mutually related input functions.** M. Robin Bai and A. L. Mielnicka-Pate (Department of Engineering Science & Mechanics, Iowa State University, Ames, IA 50011)

Statistical signal analysis approaches have been successfully used in analyzing linear and constant parameter multiple input-output systems. These methods have been particularly useful when input signals are not correlated. However, there are a number of systems that have measurable input signals that are related. An example is a vibrating plate under two excitation forces. When both the acceleration and force are monitored at a particular input, then they are correlated. In this paper three different approaches are used to investigate a two input-one output system with mutually related input signals. These approaches are: (1) conventional multiple input-output statistical signal analysis, (2) conditioned spectral analysis, and (3) a new approach, based on alternations in one of the input signals. In the last approach, the system of equations for the multiple input-output model is solved twice for two different input signal cases. The results when these three methods are used to analyze an electrical two input-one output model and a plate with two excitation forces will be presented and discussed in terms of the accuracy and required signal processing.

11:10

**MM7. Experimental study on geophysical diffraction tomography.** Tien-when Lo and M. Nafi Toksöz (E34-330, Earth Resources Laboratory, Massachusetts Institute of Technology, Cambridge, MA 02139)

Most of the diffraction tomography experiments that can be found in the literature focused on medical or NDE applications. With medical or NDE applications, these experiments allowed us to measure the scattered wave field in all desired directions. This full coverage of the targets by insonifying sonic waves, however, is impossible for objects of geophysical interests. Diffraction tomography experiments with geophysical applications have not been done yet. In order to experimentally test the applicability of diffraction tomography with geophysical applications, the authors conducted an experiment with a target immersed in water and with a source-receiver geometry similar to the source-receiver geometry of the offset VSP techniques currently being used by exploration geophysicists in the field. We measured the scattered wave field by this poor coverage but realistic source-receiver arrangement. With scattered wave field data, we reconstructed the images of our targets by a diffraction tomography algorithm. The results of our experiment are encouraging. Objects of simple geometry can be successfully imaged. In theoretical studies of diffraction tomography, most reconstruction algorithms developed so far are based on two assumptions: (1) weak scattering approximation and (2) linear behavior of scattered wave field assumption. These two assumptions impose limits on the applications of diffraction tomography and the authors took an experimental approach to delineate these limits.

11:25

**MM8. Tomographic images of submerged spheres.** Charles F. Gaumond and Phillip B. Abraham (Code 5132, Naval Research Laboratory, Washington, DC 20375-5000)

Tomographic imaging algorithms have been derived for cases of weak scattering [C. F. Schuele, H. Lee, and G. Wade, IEEE Trans. Sonics Ultrason. SU-31, 195 (1984)]. In order to study strong scattering effects on tomographic reconstructions, ideal images of rigid and elastic targets in a fluid medium will be related to known series solutions of the appropriate wave equations. Results of numerical simulations and physical experiments with a variety of spheres will also be presented.

11:40

**MM9. Correction of energy shadowing for scan images of ultrasonic data.** Doron Kishoni (NASA Langley Research Center, M.S. 231, Hampton, VA 23665)

In most of the scan image applications of ultrasonic measurements (C-scan etc.), parameters such as the maximum amplitude of the reflected waves are drawn as a function of an  $x$ - $y$  location. Function of depth may be included too. Many times the only correction that is made is the attenuation compensation, i.e., the amplitude of the reflected wave is amplified according to an exponential relation as function of depth. However, an additional parameter should be considered. This is especially important in

complex materials, such as composites, where several defects and delaminations along the ultrasonic wave path may be encountered. The parameter is the reduction in energy of the transmitted wave when a reflection from a defect occurs. Further reflections will be smaller, even if the size of the defects are the same. In this paper we address the problem and present a simple and fast algorithm that may be utilized in real-time measurements to compensate for this effect. [Work supported by NRC.]

FRIDAY MORNING, 8 NOVEMBER 1985

REGENCY BALLROOM IV, 8:30 TO 11:30 A.M.

## Session NN. Speech Communication VI: Perception of Natural and Synthetic Speech

Diane Kewley-Port, Chairman

*Hearing and Communication Laboratories, Department of Speech and Hearing Sciences, Indiana University, Bloomington, Indiana 47405*

### Contributed Papers

8:30

**NN1. Vowel errors in noise and in reverberation by hearing-impaired listeners.** Anna K. Nábělek and Paul A. Dagenais (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740)

The effects of noise and reverberation on identification of monothongs and diphthongs were evaluated using ten subjects with moderate sensori-neural hearing losses. Stimuli were 15 English vowels, spoken between /b/ and /t/ and recorded in a carrier sentence. The test was recorded without reverberation under quiet conditions then degraded either by recording in a room with reverberation time 1.2 s or by adding babble of 12 voices at speech-to-noise ratio of 0 dB. Both degradations caused statistically significant reductions of mean identification scores, but the difference between degraded means was not significant. The pattern of errors was different in noise and reverberation. Errors for monothongs in reverberation seemed to be related to an overestimation of vowel duration and to a tendency to perceive the pitch of the formant frequencies as being higher than without reverberation. Errors for monothongs in noise were not related to duration or pitch overestimation. Errors involving diphthongs were more frequent in reverberation than in noise. In reverberation, there was a tendency to judge a diphthong as the beginning monothong.

8:45

**NN2. Minimum spectral contrast for vowel identification by normal and hearing-impaired listeners.** M. R. Leck, M. F. Dorman (Department of Speech & Hearing Science, Arizona State University, Tempe, AZ 85287), and Q. Summerfield (MRC Institute of Hearing Research, University of Nottingham, Nottingham, England)

It has been suggested that the internal spectral representation of vowels in hearing-impaired listeners contains considerably less information than is available in the acoustic speech signal. It is puzzling, then, that unless the loss is quite severe, such listeners typically have little trouble identifying vowels. Apparently, faithful preservation of the vowel spectrum is not necessary for accurate identification. In this study, we sought to determine the minimum peak-to-valley differences in formant amplitude which could adequately define particular vowels for normal-hearing and hearing-impaired listeners. Vowel-like complex tones were generated by adding 30 harmonics of a 100-Hz tone in cosine phase. Amplitudes of all harmonics were equal, except for six components selected to define the formants of four vowels, which listeners were asked to identify. The selected components varied relative to the remaining component amplitudes by +1 to +8 dB. Results indicated that the 2–3-dB peak-to-valley contrast required for identification by normal hearing listeners had to be doubled

and tripled, respectively, for normal listeners in noise and for hearing-impaired listeners. While the spectral contrast information in vowels is more than sufficient for identification by the hearing-impaired, these listeners are operating closer to the limits of correct identification than their normal-hearing counterparts. This procedure offers a paradigm for a finer quantification of differences in vowel perception by these two populations than the usual accuracy measures. [Research supported by NIH.]

9:00

**NN3. Relevance of time-varying properties of the first formant frequency in vowel representation.** Maria-Gabriella Di Benedetto<sup>1</sup> (Research Laboratory of Electronics, Room 36-529, Massachusetts Institute of Technology, Cambridge, MA 02139)

Acoustic analysis of the vocalic portions of CVC nonsense syllables spoken by three speakers (two males and one female) in the sentence frame "The—again" shows that ambiguities between vowels, for each speaker, occur if the vowels are represented by the values of  $F1$  and  $F2$  sampled at the time where  $F1$  reaches its maximum. These ambiguities occur primarily in the  $F1$  dimension. The examination of the  $F1$  trajectories of the vowels for which confusion occurs, shows variations in the initial slope of this trajectory among different vowels. In particular, if two different vowels such as /t/ and /ε/ have the same maximum  $F1$ , then  $F1$  for the lower vowel reaches its maximum value earlier. Perceptual experiments have been carried out to examine the perceptual importance of  $F1$  slope, using synthetic CV syllables. Preliminary results are in agreement with the hypothesis that stimuli with a steeper initial slope of  $F1$  are perceived as lower vowels. Results are similar for subjects of different languages, leading to a suggestion that this phenomenon can be explained on an auditory basis. <sup>1</sup> On leave from Department of Information and Communication, University of Rome, La Sapienza, Italy.

9:15

**NN4. The perceptual reality of a formant frequency.** Dennis H. Klatt (Room 36-523, Massachusetts Institute of Technology, Cambridge, MA 02139).

A series of vowels similar to [i] was synthesized using a different constant value of fundamental frequency ( $f_0$ ) for each member. As  $f_0$  ranged from 133 to 200 Hz, harmonics either fell exactly at the first formant frequency ( $F1$ ) or a pair of harmonics straddled  $F1$  with varying degrees of skew and thus differing relative amplitudes. The presumed task of the auditory system is to interpret harmonic amplitudes in the neighborhood of  $F1$  in order to estimate the true  $F1$  frequency of the vowel. It will first be

shown that (1) energy-based methods of interpreting filter-bank outputs, (2) non-pitch-synchronous linear prediction analysis, and (3) synchrony measures based on auditory models all systematically mis-estimate  $F1$  frequency by as much as plus/minus 10% as  $f_0$  varies. Secondly, a perceptual experiment was performed to determine whether these systematic estimation errors are present in auditory judgement data. Comparison of the standard stimulus series with one in which  $F1$  is "corrected" so that a filter bank sees a constant  $F1$  indicates that the uncorrected fixed- $F1$  series has more nearly constant phonetic quality. Accounting for the perceptual data in terms of simple processing strategies is not yet possible, but if the central auditory system is provided with the frequencies and amplitudes of harmonics near  $F1$ , a moderately complex calculation could yield  $F1$  with minimal error. [Work supported in part by an NIH grant.]

9:30

**NN5. Vowel perception in noise.** J. Besing, R. R. Hurtig, and M. J. Collins (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Vowel confusion data presented by Pickett [J. Pickett, J. Acoust. Soc. Am. 29, 613-620 (1957)] were reanalyzed using a multidimensional scaling procedure. The data were analyzed using this method to determine the relative contribution of each dimension under different noise and context conditions. The original data were collected using vowels produced in phonetically balanced words and bVb syllables under different noise spectra conditions. Data were fit with a three-dimensional solution; two dimensions were comparable to  $F1$  and  $F2$ . The subject weights provided estimates of the relative salience of each dimension for each noise condition. Examination of those weights indicated that the relative salience of each dimension varied with vowel context. Dimensions one and three were more heavily weighted for bVb contexts. Dimensions two and three were weighted more heavily for phonetically balanced contexts. Differences in weighting of each dimension yielded perceptual representations that were distortions of the traditional representation of vowel space in terms of  $F1$  and  $F2$ .

9:45

**NN6. Nonlinear auditory coding of vowel formants at high sound pressure levels.** M. F. Dorman, J. M. Lindholm, M. T. Hannley, and M. R. Leek (Departments of Speech and Hearing Science and Psychology, Arizona State University, Tempe, AZ 85287)

To study vowel intelligibility in the absence of the acoustic reflex, ten synthetic vowels whose duration was shorter than the effective latency of the acoustic reflex were presented to normal hearing listeners at levels ranging from 72 to 106 dB SPL. Signal intelligibility varied as a function of both SPL and vowel identity. Vowels with widely spaced formants, i.e., /i/ and /ɪ/, were unaffected by presentation level. Vowels with more proximal formants began to show a decrement in intelligibility at 96 dB and suffered as much as a 40% decrement at 106 dB. Confusion errors were not symmetrical in  $F1$ - $F2$  space suggesting a systematic distortion in the auditory coding of formant location. These data indicate another source of distortion for many severely hearing-impaired listeners who commonly listen to speech amplified to greater than 100 dB and who do not have measureable acoustic reflexes. [Research supported by NIH.]

10:00

**NN7. Relationship between LP-residual spectral distances and phonetic judgments.** C. Kamm and D. Kahn (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

The relationship between LP-residual spectral distances [F. Itakura, IEEE Trans. Acoust. Speech Signal Process. ASSP-23, 67-72 (1975)] and phonetic judgments was examined in two studies. One analysis compared published phonetic similarity results [D. Klatt, Proc. ICASSP, 1278-1281 (1982)] with Itakura distances between steady-state synthetic reference vowels (/æ/ and /a/) and a set of acoustic variants of each vowel (approximating Klatt's stimulus sets). Despite the documented utility of the Itakura measure for speech recognition, correlations between the phonetic similarity measures and Itakura distances were very low (0.20 and

0.02 for /æ/ and /a/, respectively). Significant differences in rank across the two measures were observed for acoustic variants incorporating spectral tilt, low-pass filtering and changes in  $F3$ . A second experiment measured Itakura distances and phonemic identification judgments for 21 acoustic variants of each of ten English reference vowels. For six reference vowels, at least one variant was identified as phonemically distinct from its reference: these phonemically distinct variants yielded *smaller* Itakura distances than 48% of those variants judged *not* phonemically distinct from the reference. These results quantify and corroborate the well-known lack of optimality of the Itakura distance measure for speech recognition and may contribute to the design of more appropriate distance metrics.

10:15

**NN8. Acoustic and perceptual correlates of anticipatory lip rounding.** F. Bell-Berti (Department of Speech, Communication Sciences, and Theatre, St. John's University, Jamaica, NY 11439 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), C. E. Gelfer, and K. S. Harris (Graduate School, CUNY, New York, NY 10036 and Haskins Laboratories, New Haven, CT 06511)

Studies of coarticulation have provided evidence of lip rounding activity during consonants immediately preceding a rounded vowel, although reports differ as to the extent of this anticipatory activity. One would naturally expect to find this rounding reflected in the acoustic spectra of the consonants, and to have perceptual salience. To establish the extent of such acoustic and perceptual effects, we conducted studies using two speakers, one who uses lip gestures for alveolar consonants in both rounded and unrounded vowel environments, and another who uses them only in rounded vowel environments. In the first study, spectral analyses of 520  $V_1C_nV_2$  utterances from two speakers yielded little evidence of  $V_2$  rounding gestures effecting an unrounded  $V_1$ , while there is evidence that such lip rounding gestures influence fricative noise immediately preceding a rounded or unrounded  $V_2$ . The results indicate that the acoustic effect is not perceptually salient for a speaker for whom lip rounding is inherent in alveolar consonant production. [This work was supported by grant NS-13617 and RR-05596.]

10:30

**NN9. Acoustic and perceptual correlates of prevocalic pharyngeal and uvular consonants in Arabic.** Abeer Alwan (Research Laboratory of Electronics and the Department of Electrical Engineering and Computer Science, Room 36-529, Massachusetts Institute of Technology, Cambridge, MA 02139)

The first part of this study investigated the acoustic correlates of two pharyngeal consonants /ʕ,ħ/ and three uvular consonants /χ,ħ,ʁ/ in prevocalic position with the three vowels /a/, /i/, and /u/, in Arabic. Analysis of these consonants showed distinctive formant trajectories for each class of sounds and the existence of several allophonic realizations for the voiced consonants in both classes. The second part of the study examined the perceptual correlates of the two voiced consonants /ʕ/ and /χ/ prevocalically with the vowel /a/. Formant trajectories and bandwidths were manipulated independently in synthetic nonsense /Ca/ syllables. These synthetic stimuli were then presented to subjects in identification tests. Results show that a high  $F1$  and the proximity of  $F1$  and  $F2$  are essential for the perception of /ʕ/, whereas a widened  $F1$  bandwidth is essential for the perception of a natural /χ/. The acoustic and perceptual results are discussed in terms of the mechanisms of production of sounds with a narrow constriction between the glottis and the velum, and the corresponding articulatory-acoustic transformations involved. [Work supported in part by an NSF grant.]

10:45

**NN10. Perceptual learning of synthetic words and sentences.** Steven L. Greenspan, Howard C. Nusbaum, and David B. Pisoni (Speech Research Laboratory, Psychology Department, Indiana University, Bloomington, IN 47405)

Previous research has shown that the intelligibility of synthetic speech can be improved with training [E. C. Schwab, H. C. Nusbaum, and D. B. Pisoni, *Human Factors* (in press)]. In the present study, we investigated the relationship between the type of training subjects receive and the pattern of perceptual learning that occurs. Three groups of subjects were given word recognition tests consisting of synthetic words and sentences generated by the Votrax Type-n-Talk, before and after a training period. One group of subjects was trained on isolated synthetic words. A second group was trained on fluent synthetic sentences. A control group received no training at all. Recognition of isolated synthetic words improved equally for both groups trained on synthetic speech compared to the control group. However, overall word recognition in sentences only improved for subjects that were trained on synthetic sentences. These results demonstrate that performance with synthetic sentences will predict performance with isolated words, but the converse may not always hold. [Work supported by AFOSR and NIH.]

11:00

**NN11. Perceptual attention in monitoring natural and synthetic speech.** Howard C. Nusbaum, Steven L. Greenspan, and David B. Pisoni (Speech Research Laboratory, Psychology Department, Indiana University, Bloomington, IN 47405)

The role of voice distinctiveness and phonetic discriminability in perception of natural and synthetic speech was investigated. Subjects were instructed to monitor sequences of CV syllables for a specified target syllable in several conditions: (1) targets and distractors produced by the same human talker ( $N/N$ ); (2) targets produced by a synthetic talker and distractors produced by a human talker ( $S/N$ ); and (3) targets produced by a synthetic talker and distractors produced by both the same synthetic talker and a natural talker ( $S/N + S$ ). Results indicate that highly intelligible synthetic targets are detected faster mixed with natural distractors, than are natural targets mixed with natural distractors. However, when sub-

jects are required to discriminate between synthetic targets and synthetic distractors, performance is much worse than for natural targets and natural distractors. The distinctive mechanical sound of synthetic speech only appears to aid perception when there is just a single synthetic message among natural messages. When listeners must discriminate among synthetic messages, performance is significantly worse than when they must discriminate among natural messages. [Work supported by AFOSR and NIH.]

11:15

**NN12. Consonant confusions and perceptual spaces for natural and synthetic speech.** Moshe Yuchtman, Howard C. Nusbaum, and David B. Pisoni (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Earlier research in our laboratory has demonstrated that synthetic speech is less intelligible and more capacity demanding than natural speech. These differences appear to be related to processes responsible for encoding the input signal into a segmental phonemic representation. There are several hypotheses that could account for the greater difficulty involved in synthetic speech perception. One hypothesis is that synthetic speech is structurally equivalent to natural speech degraded by noise. An alternative hypothesis is that the acoustic-phonetic structure of synthetic speech is impoverished in comparison to natural speech in that a minimal set of acoustic cues are used to implement phonetic segments. The two hypotheses lead to different predictions about the nature of synthetic consonant confusions in relation to confusions of natural speech degraded by noise. To resolve this issue, we carried out multidimensional scaling analyses of confusion matrices for synthetic consonants produced by DEC-Talk, Prods-2020, and the Votrax Type-N-Talk, and for natural speech at several S/N ratios. The results of the analyses tend to support the second hypothesis: The properties of the perceptual spaces obtained for the synthetic consonants differed considerably from those obtained for the natural consonants. [Work supported by AFOSR.]

FRIDAY MORNING, 8 NOVEMBER 1985

REGENCY BALLROOM II AND III, 9:00 A.M. TO 12 NOON

## Session OO. Education in Acoustics: Personal Computers for Laboratory Instruction

Murray S. Korman, Chairman

*Physics Department, U. S. Naval Academy, Annapolis, Maryland 21402*

Chairman's Introduction—9:00

### Invited Papers

9:03

**OO1. Applications of the Apple computer for classroom demonstrations and laboratory experiments.** Lee N. Bolen and Lawrence A. Crum (Physical Acoustics Research Laboratory, The University of Mississippi, University, MS 38677)

The Apple computer game port can be used as a timing circuit for inexpensive interactive demonstrations and laboratory experiments. Examples of the use of this technique to measure the velocity of a baseball thrown in the classroom by the most adept students as well as the measurement of speed and acceleration of a slot car racer in the laboratory will be demonstrated.

9:21

**OO2. The use of microcomputers in an undergraduate acoustics laboratory.** Murray S. Korman (Department of Physics, U. S. Naval Academy, Annapolis, MD 21402)

Microcomputers are used in the laboratory as a tool to collect, store, analyze, and graph data from acoustics experiments. Emphasis is placed on teaching the physics involved from individual experiments and not on

particular features of the computer. These microstations allow students to record data from the experimental apparatus using the computer. Simple menued software instructions are available to organize the collection of data for printout or for  $X$ - $Y$  pairs to be plotted and labeled. Students can also scale their data and store or plot these variations. This laboratory facility speeds up the tedious and routine data taking chores of the experiment and frees up time for discussion of the physics. In many cases time allows the student a chance to try a few things on his own. Simple acoustics experiments will be shown involving transient vibration and Fourier analysis using our mini-computer station. This station has a 6502 microprocessor, a two-channel analog to digital plug-in board, a disk drive, monitor, and printer. Use is made of an inexpensive FFT program. Currently, there are four stations available in the acoustics course taught for physics and engineering majors. [Work supported by USNA.]

9:39

**OO3. Microcomputer as a laboratory instrument in the basic physics lab.** R. Shelby, D. Nordling, and D. Sadler (U. S. Naval Academy, Annapolis, MD 21402)

The microcomputer-based measuring station is proving to be a versatile, high-performance tool for use in data gathering, plotting, storage, and analysis in undergraduate physics laboratories. This paper will discuss the large scale use of microcomputer-based measuring stations at the Naval Academy during the last year. Topics considered will include our reasons for changing to measuring stations and reasons for specific equipment choice, our experience in the laboratories with over 1000 students, necessary elements for basic and more advanced stations, the requirement for simple but versatile software, and the philosophy necessary to keep the focus of the laboratories on physics rather than computer science. The specifics of the equipment and software used in our program will be reviewed and a brief look at the future will be given.

9:57

**OO4. Microcomputers and the undergraduate laboratory: An ideal match.** Howard Saltsburg, Richard Heist, and Thor Olsen (Department of Chemical Engineering, University of Rochester, Rochester, NY 14627)

Making the undergraduate laboratory a useful educational experience requires that the tedious data acquisition and reduction of raw data be minimized in favor of enabling the student to concentrate on the scientific and engineering aspects of the study. The use of the microcomputer not only permits that approach to be implemented but offers new opportunities for improved laboratory studies. For the past 5 years, we have been engaged in the introduction of the microcomputer into the undergraduate chemical engineering laboratories. Primary emphasis was placed upon improving the technical content of the program. To provide for efficient operation of the laboratory, every computer-aided experiment has a dedicated computer. This required that we utilize inexpensive computers and minimize the cost of the interfaces needed for appropriate experimental variables. In no sense, however, has a computer laboratory been developed. The computer is simply a tool and only those studies which can profit from the use of the computer have been interfaced. A brief description of the development of the program will be given, several examples of laboratory studies will be described, and some experiments will be demonstrated.

10:15

**OO5. Use of Commodore computers in the undergraduate mechanical engineering instrumentation laboratory course.** Roger L. T. Oehmke and William J. Wepfer (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Integration of microcomputers into a required junior-level mechanical engineering course resulted during a recent modernization of the undergraduate program. The course presents the microcomputer as a powerful laboratory instrument (tool) and follows the approach taken by Saltsburg, Heist, and Olsen [MICRO 53, 53-55 (October 1982); 55, 59-63 (December 1982); 56, 38-43 (January 1983); 57, 89-91 (February 1983)] at the University of Rochester. The emphasis is on the applications of the computer as a flexible data acquisition and control device. After completion of a 3-h introductory laboratory the students undertake a 6-h lab exercise in which the computer is used to monitor and control the RPM of a dc motor. Remaining experiments focus on traditional mechanical measurements, many involving use of the computer. An example is the use of a digital oscilloscope peripheral to collect vibration data (acceleration versus time) from an accelerometer mounted on one of two bearing blocks that support a motor-driven rotating shaft.

10:33

**OO6. Microcomputer based data acquisition and analysis systems for graduate student research in Acoustics.** Oliver H. McDaniel, John S. Lamancusa, William C. Ward, and Kevin Todd (Department of Mechanical Engineering, The Pennsylvania State University, University Park, PA 16802)

The development and commercial availability of dedicated multichannel fast-Fourier transform (FFT) analyzers have resulted in a significant advancement in productivity in research in acoustics. The capability of at least two-channel FFT instrumentation is a must for any acoustics laboratory; however, the acquisition of a single instrument represents a major item in a research budget. Two personal-computer-based multichannel FFT systems will be described and demonstrated. These systems are currently being used by graduate students



in these research in internal combustion engine noise and in piping system noise. These relatively low-cost systems were assembled with commercially available data acquisition components and software. Microphone signal conditioning preamplifiers and anti-aliasing filters were designed by a senior electrical engineering student. Preamplifier gains and filter cutoff frequencies are controlled by the microcomputer.

### Contributed Papers

10:51

**OO7. Waveform synthesis on a personal computer.** George Knott (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

An inexpensive music synthesis program (*Concertware*, by Great Wave Software) is shown to be an alternative to traditional analog methods in the teaching of waveform synthesis concepts in the laboratory. In addition, the pedagogical ramifications of using a streamlined software package such as *Concertware* in what can best be called the "computer laboratory" versus the traditional equipment intensive lab environment are discussed.

11:03

**OO8. A personal computer workstation for music education.** Mark Dolson (Computer Audio Research Laboratory, Center for Music Experiment, Q-037, University of California, San Diego, CA 92093)

Courses in acoustics and psychoacoustics are becoming increasingly common within the music curriculum. An introduction to these subjects enables students both to better understand the structure of traditional music and to explore contemporary alternatives to that structure. But the information in these courses is generally imparted to students exclusively by means of lecture/demonstration. This mode of presentation makes the material remote and difficult to absorb. A sufficiently powerful personal computer with analog-to-digital and digital-to-analog converters can overcome this problem by providing the student with a rich environment in which to explore the material directly. In this report we describe a major software-development project aimed at producing a coherent personal-computer-based workstation for sound analysis, synthesis, and modification. The software is written in C and intended for a UNIX environment. Much of it runs at present on a VAX 11-780 and is used for computer music composition and research. The package currently under development, though, has education as its primary focus. Since this development may continue for some time to come, an important goal of this report is to solicit informed opinions and contributions from a diversity of sources.

11:15-12:00

Demonstration Period

FRIDAY MORNING, 8 NOVEMBER 1985

DAVIDSON ROOM B, 9:00 TO 11:00 A.M.

### Session PP. Physical Acoustics VIII: Propagation and Waveguides

Pieter S. Dubbledday, Chairman

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

### Contributed Papers

9:00

**PP1. Fiber-matrix mode acoustic wave propagation in carbon-epoxy composites.** William T. Yost and John H. Cantrell (NASA-Langley Research Center, Mail Stop 231, Hampton, VA 23665)

We report the generation and detection of coupled fiber-matrix mode acoustic wave propagation along the fiber direction in eight-ply unidirectional carbon-epoxy composites using a surface generation and detection mechanism. Compressional, 0.8- to 1.2-MHz acoustic waves are transmitted and received through transducer angle blocks placed on the sample surface. The angles are adjusted such that the component of the wave propagation vectors in the angle blocks along the sample surfaces (hence, along the fiber direction) equals the component of the wave propagation vector in the composite along the fiber direction (Snell's law). The measured sound velocity of  $(9.29 \pm 0.04) \times 10^3$  cm/s at 1 MHz using this technique does not correspond to the values obtained for wave propagation in either the fiber or the matrix alone. A model based on the law of mixtures together with the assumption of hexagonal symmetry for the unidirectional composite shows that the wave propagates as a coupled fiber-matrix

mode along the C-axis direction. The model predicts a value of the sound velocity in such a mode that is consistent (to within experimental error) with that obtained from the experiments.

9:15

**PP2. Effect of pore fluid viscosity on the velocities of acoustic waves in porous rocks.** Zhijing Wang and Amos Nur (Department of Geophysics, Stanford University, Stanford, CA 94305)

Compressional and shear velocities of acoustic waves propagating in porous rocks saturated with viscous fluids were measured both in the kHz and MHz frequency range. The experimental results of the velocities were plotted as a function of pore fluid viscosity ranging from 1 to  $10^{10}$  centipoise. It was shown that the measured velocities increased with increasing pore fluid viscosity in the kHz frequency range, while in the MHz range, the pore fluid viscosity did not have much effect on the acoustic wave velocities. The experimental results were also discussed in terms of theories of wave propagating in viscous fluids and in the saturated porous so-

lids. Applications of the results to oil production and exploration, as well as to the low velocity zone in the earth, were also presented.

9:30

**PP3. Sound speed in air: Past and present.** George S. K. Wong (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

The calculation of a new value for the sound speed ( $C_0 = 331.29 \text{ ms}^{-1}$ ) in standard dry air at  $0^\circ\text{C}$  and at a barometric pressure of 101.325 kPa is presented. The maximum uncertainty in this sound speed  $C_0$  is estimated to be approximately 200 ppm. The theory of the above calculation is based on the equation of state, and with the knowledge of  $\gamma/M$  which is derived from published theoretical and experimental thermodynamic data on the constituents of the standard atmosphere [G. S. K. Wong and T. F. W. Embleton, *J. Acoust. Soc. Am.* **76**, 555–559 (1984)], and  $M$  is the molar mass. Investigations which led to the general acceptance of the previous sound speed are examined, and there is strong evidence that enables one to conclude that the maximum possible uncertainties in previous sound-speed assessments are sufficient to encompass the above new sound speed. The effects of carbon dioxide on sound speed are discussed.

9:45

**PP4. Analogies between nonflat ground and nonuniform meteorological profiles in outdoor sound propagation.** Tony F. W. Embleton (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

The analogy can be made between downwardly curving ray paths (as in a temperature inversion) over flat ground and an upwardly curving ground (a rising hillside) in a neutral atmosphere. Conversely, temperature lapses and flat ground are analogous to a neutral atmosphere and falling ground. The change in sound levels relative to those at the same range either in free space or over a flat, hard ground in a neutral atmosphere can be predicted by considering the appropriate ray paths or wave components involved. In all cases the change in sound level is a function of frequency. In general, sound levels increase during a temperature inversion or up a rising hillside, and decrease during a temperature lapse or down a falling hillside. Some field measurements exist to check the general structure of these predictions, but not many at ranges of several kilometers, and there is a need for verification at a variety of topographical sites under various meteorological conditions.

10:00

**PP5. Measurement of normal-incidence impedance of outdoor ground surfaces in the frequency range 20–500 Hz.** G. A. Daigle and M. R. Stinson (Division of Physics, National Research Council, Ottawa K1A 0R6, Canada)

The impedance of natural ground surfaces has been measured extensively at frequencies above about 200 Hz. At lower frequencies, however, measurements are scarce and have been inaccurate. We have begun using a two-microphone, phase-difference technique similar to that of Nicolas and Legouis [Proc. 2nd Int. Congress on Acoustic Intensity, Senlis (1985)], which promises to overcome some of the difficulties at low frequencies. A point source is suspended above the ground and the sound field is measured with two phase-matched microphones along the vertical line below the source. The reflection coefficient, and hence the impedance, is obtained from the variation of the phase difference between microphones as a function of height above the ground surface. The use of two closely spaced microphones has the advantage of minimizing the effects of random fluctuations due to turbulence. Preliminary measurements have been made down to 50 Hz over grass covered ground. Further measurements are underway in the range 20–500 Hz.

10:15

**PP6. Ray, mode, and hybrid options for time-dependent source-excited vibrational motion in an elastic plate.** I. T. Lu and L. B. Felsen

(Department of Electrical Engineering and Computer Science, Polytechnic Institute of New York, Route 110, Farmingdale, NY 11735)

A hybrid theory for source-excited propagation in multiwave multi-layer media is developed whereby ray fields and normal mode fields (with a smoothing remainder) are combined in self-consistent proportions so as to take advantage of the favorable features of each of these descriptions [I. T. Lu, L. B. Felsen, and A. H. Kamel, *Wave Motion* **6**, 435–457 (1984)]. To avoid proliferation of multiple reflected ray fields, caused by wave coupling at boundaries, a new spectral object called “eigenray” has been introduced, which has dispersive characteristics similar to those of a normal mode but undergoes reverberations like a single ray field in a single wave medium. For time-harmonic excitation by a line forcing function, these new formulations have already been tested numerically on the simple but nontrivial example of  $P$ -SV propagation in a single elastic plate [I. T. Lu and L. B. Felsen, *J. Acoust. Soc. Am.* (to be published)]. The validity of the hybrid algorithm has been confirmed, and parameter regimes have been found wherein the hybrid approach offers a competitive alternative to other options. The test calculation is extended to excitation by a high-frequency Gaussian pulse transient source. The results reveal the complicated multiple arrival structure, well resolved at early times but tending toward modal behavior at later times. These features, which pertain especially to long observation intervals and wave constituents with high-frequency spectral content, are explained well, and are computed efficiently, by the “optimal” hybrid format. [Work supported by NSF and ONR.]

10:30

**PP7. Normal mode propagation in a rough-walled waveguide.** Gerald L. D'Spain,<sup>(1)</sup> Ken J. Reitzel,<sup>(1)</sup> and Herman Medwin<sup>(2)</sup> (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The effects of a randomly rough boundary on normal mode propagation in a rigid-walled waveguide have been studied theoretically and experimentally. The theoretical solution considers steep-sloped roughness elements and uses the low-frequency technique of Tolstoy [*J. Acoust. Soc. Am.* **75**, 1–22 (1984)]. It is predicted that, at frequencies below the first mode cutoff, the mode generated in the rough-walled waveguide has a greater amplitude at the rough boundary than at the smooth boundary and propagates with a constant phase plane wavefront at a slower speed than in the smooth-walled waveguide. The laboratory experiment, using gravel and frequencies up to 10 kHz, shows excellent agreement with the theoretical prediction. The eigenfunction perturbations and phase velocity changes caused by steep-sloped roughness elements are discussed for higher frequencies and higher order modes. [Research supported by the Office of Naval Research.] <sup>(1)</sup>Ocean Acoustics Associates, Pebble Beach, CA 93953.

10:45

**PP8. Strain sensing using interface waves in clad rod acoustic waveguides.** Richard O. Claus and Kimberly D. Bennett (Department of Electrical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

The measurement of strain in materials using clad rod acoustic waveguides has been considered recently by several authors [R. T. Harrold and Z. N. Sanjana, *Proc. Soc. Plastics Eng. Conf.*, Washington, DC (May 1985)]. In such waveguides, axisymmetric torsional, axisymmetric radial-longitudinal, and core-guided shear modes typically propagate if the velocity of plane shear waves in the clad exceeds that velocity the core. If instead the materials are reversed, these modes are not supported but an interface wave may propagate on the core-clad boundary. In principle, such waves may propagate without attenuation if the core and clad materials properties are suitably related. The theoretical sensitivity of the attenuation and mode conversion of such nonattenuating interface waves to strain-induced variations in the geometry and elastic constants of the waveguide materials are considered in this paper. Experimental measurements using strained glass-on-glass clad rods which approximately satisfy the assumed boundary conditions are reported. Potential applications in the internal monitoring of materials are suggested. [Work supported by NASA and Simmonds Precision.]

## Session QQ. Shock and Vibration III: Structural Vibration; Analysis and Control

Louis A. Herstein, Chairman  
*Naval Sea Systems Command, Washington, DC 20362*

Chairman's Introduction—9:00

*Contributed Papers*

9:05

**QQ1. Vibration response of ring-supported, fluid-loaded, circular cylindrical shells to turbulent boundary layer pressure fluctuations.** David A. Bostian (Naval Underwater Systems Center, New London, CT 06320) and Courtney B. Burroughs (Applied Research Laboratory, The Pennsylvania State University, P.O. Box 40, State College, PA 16804)

Using the analytic model presented at the 109th Meeting of the Acoustical Society of America at Austin, TX in April 1985, predictions of the vibration response of ring-supported, fluid-loaded, circular cylindrical shells to turbulent boundary layer pressure fluctuations are presented. At different frequencies, wavenumber spectra of the shell response are presented and wavenumber bands where the shell response is maximum are identified. Examples of the dependence of the shell response on the observation location, and ring size and spacing are also given. Sums over wavenumbers are taken to obtain the response of the shell at a point as a function of frequency.

9:20

**QQ2. Determining the spatial decay rate of free waves in layered plates.** Paul W. Jameson and Khushi Chandiramani (Department 8C, BBN Laboratories, 10 Moulton Street, Cambridge, MA 02238)

We will show that the spatial decay rate of free bending waves in multilayered fluid-loaded plates can be estimated very easily using only a knowledge of the input admittance of the plate as a function of the frequency  $\omega$  and wavenumber  $k$  of the normal stress applied to the plate. If the admittance is written in the form of a magnitude and a phase  $\theta$ , the spatial decay rate  $\alpha_s$  for free waves is given approximately by the expression

$$\alpha_s(\omega) \approx -2 \left/ \frac{\partial \theta(\omega, k)}{\partial k} \right|_{\theta=0}$$

The evaluation of the partial derivative of  $\theta$  is performed with respect to real values of  $k$ . The justification for the result is based on arguments by K. Chandiramani and G. B. Witham which identify a proportionality between the Lagrangian energy density and the derivative with respect to frequency of the imaginary part of the surface impedance of the plate.

9:35

**QQ3. Random superharmonic resonance.** Srivatsa Rajan and Huw G. Davies (Department of Mechanical Engineering, University of New Brunswick, Fredericton, New Brunswick, Canada E3B 5A3)

The third-order superharmonic mean-square response of a Duffing oscillator to narrow-band random excitation is analyzed. The analysis uses harmonic balance to separate the primary and superharmonic responses and statistical linearization to obtain a differential equation for the autocorrelation of the superharmonic response. The main emphasis of the analysis is to demonstrate the effect of excitation bandwidth on the response and stability of the oscillator. The analysis shows that a tripling of both excitation frequency and bandwidth occur and that occurrence of multivalued superharmonic response and jump phenomena depends upon the bandwidth of excitation. Stability of the superharmonic response is

considered by generating a mean-square phase plane which shows that multiple values represented by two stable sinks and an unstable saddle point occur when the excitation bandwidth is small. The phase plane reduces to a single sink when the excitation bandwidth is increased. Stability is also examined by perturbing the superharmonic response equation. The locus of vertical tangents and stability regions are obtained. The theoretical results are compared with results obtained from digital simulation. [Work supported by NSERC, Canada.]

9:50

**QQ4. Variance in magnitude and phase of structural transfer functions.** Richard H. Lyon and Robert Gibson (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

The statistics of transfer functions has long been of interest in room acoustics because of the necessity of frequency averaging to generate smooth response curves. An important parameter in these statistics is modal overlap, the ratio of modal bandwidth to modal spacing. When this parameter is large, as it often is in room response, the magnitude statistics become fairly simple. The phase statistics in room response have not been greatly studied. Time delays in room acoustics are more often related to pulse arrival statistics using an image model for wall reflections. Structures are different in the behavior of their transfer functions primarily because they tend to have low modal overlap. This means that both the magnitude and phase of their transfer functions need to be reevaluated. This has been done and the results are presented in this paper. Of particular interest are the phase statistics, which appear to be derivable from a random walk model.

10:05

**QQ5. A comparison of optical and numerical methods for analyzing structural vibrations.** Tammy Evans (David Taylor Naval Ship Research and Development Center, Bethesda, MD 20084) and Joseph A. Clark (Mechanical Engineering Department, Catholic University of America, Washington, DC 20064)

An optical holographic interferometry method has been used to determine the resonant frequencies, mode shapes, and peak displacement amplitudes of an irregular, clamped, vibrating plate. The plate was driven by an electromagnetic shaker with swept sine wave execution and resonances were observed by time holographic interferometry. Mode shapes at several resonant frequencies were then photographed by a time-average holographic interferometry method. The observed resonant shapes were compared with predictions obtained by two finite element programs (GIFTS and NASTRAN). Facilities required for each method of solution and procedures for collecting and interpreting mode shape data will be described. Applications of the optical and numerical methods to problems involving damped vibrating structures will be discussed. [Research supported by DTNSRDC and ONR.]

10:20

**QQ6. The use of finite-element techniques in statistical energy analysis.** Richard N. Brown (BBN Laboratories, 10 Moulton Street, Cambridge, MA 02238)

When using the statistical energy analysis (SEA) technique to compute the structureborne and acoustic power flow through complex structures, a parameter of paramount importance is the coupling loss factor which characterizes the power flow across substructure boundaries. For many simple boundary types (e.g., plates connected to plates) the coupling loss factors are available in the literature, but in applying the SEA method to structures with more complex connections the finite element method can be used to advantage in computing these factors. The structure studied here consists of two beams, capable of supporting flexural, compressional, and torsional motion, connected by a small, builtup box which has several resonances in the frequency range of interest. The admittance matrix of the box structure was computed using NASTRAN. From this admittance matrix the coupling loss factors are computed. The resonant structure of the "coupler" (the box) is apparent in the results, which compared well with results from experiment.

10:35

**QQ7. Experiments on reduction of aircraft interior noise using active control of fuselage vibration.** Chris R. Fuller (Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

Interior sound in aircraft fuselages is directly coupled to fuselage vibration. This suggests the use of active vibration control of the fuselage vibration to reduce interior noise levels. In this paper, preliminary experiments to investigate this new technique are discussed. The experiments were performed on a closed cylindrical shell excited by an external acoustic source, representative of an aircraft fuselage. Active vibration control of the shell was achieved by a single minishaker attached pointwise to the exterior wall of the cylinder. Sound levels throughout the cylinder were measured by a moveable microphone traverse. The interior levels were found to be attenuated at most locations between 10 and 35 dB by the active vibration control with a constant control amplitude and phase, i.e., global attenuation was achieved. The physical mechanisms behind the effect are discussed. The new method shows much potential for reduction of propeller interior noise in aircraft, without the penalty of large added weight.

10:50

**QQ8. Vibration isolation of a bus driving lane in a multipurpose structure.** Dana S. Hougland (David L. Adams Associates, Inc., 1701 Boulder Street, Denver, CO 80211)

A 19 500 square foot, vibration-isolated, bus driving lane was designed and installed above grade in a multifunction structure. The isolated driving lane was designed to prevent structural fatigue and vibrational annoyance in a high-rise office tower erected on the same substructure. A criterion for bus-induced acceleration in the structure adjoining the bus driving lane was established at a maximum of  $-55$  dB (re:  $1.0g_{rms}$ ) in the 10- to 20-Hz frequency range. The isolation system was designed in separate slab sections using removable springs with neoprene dampening elements. Edge and end conditions were treated at each slab section to allow smooth transfer between slab sections. Porting was provided beneath the slab to minimize the effects of air viscosity and the additional dynamic stiffness of entrapped air. The resonant frequencies and mode shapes of each slab section were studied using finite element analysis techniques. Preliminary test results on the vibration-isolated bus driving lane and adjoining structure are presented.

11:05

**QQ9. Resonant frequencies and mode shapes of open prolate spheroidal shells.** Jack Yahner and Courtney B. Burroughs (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

Governing differential equations are derived for an open prolate spheroidal shell with inner and outer surfaces defined by prolate spheroidal coordinate surfaces. Transverse shear and rotary inertia are included in the shell equations. The open end of the shell, defined by an axisymmetric coordinate line, is assumed clamped. Approximate solutions for the axisymmetric resonant frequencies and mode shapes of the *in-vacuo* shell are derived using Galerkin's variational method.