

PROGRAM OF

The 114th Meeting of the Acoustical Society of America

Hyatt Regency Hotel • Miami, Florida • 16–20 November 1987

MONDAY EVENING, 16 NOVEMBER 1987

UM AUDITORIUM, 7:00 TO 9:00 P.M.

Tutorial Lecture

Introduction to Musical Acoustics. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Musical acoustics is a broad interdisciplinary field that deals with the production, transmission, and perception of musical sound. It overlaps several other branches of acoustics, such as physical acoustics, shock and vibration, architectural acoustics, psychoacoustics, speech communication, and electroacoustics. In this introductory tutorial, considerable attention will be paid to how sound is generated by various musical instruments. In percussion instruments and in certain string instruments, the player supplies energy by striking or plucking the primary oscillator (string, bar, membrane, or plate), which in turn transmits energy to other parts of the instrument. Wind instruments, on the other hand, depend upon nonlinear feedback from the air column to the input control valve in order to generate sustained oscillations. Similar feedback exists in a bowed string instrument, and to a lesser extent in the singing voice. Time will allow only a brief discussion of how the sound is transmitted from the source to the listener via the auditorium or by electroacoustic means, such as recording, reproducing, and broadcasting systems. Finally, certain aspects of the perception of musical sound (such as loudness, pitch, and timbre) will be considered.

TUESDAY MORNING, 17 NOVEMBER 1987

BOUGAINVILLEA ROOM, 8:30 TO 11:05 A.M.

Session A. Engineering Acoustics I: Structure Interaction and Transducer Arrays

Anthony J. Rudgers, Chairman

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

Chairman's Introduction—8:30

Contributed Papers

8:35

A1. Application of a variational principle for fluid–structure interaction to the analysis of the response of an elastic disk in a finite baffle. Jerry H. Ginsberg and Pei-Tai Chen (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A variational principle derived from the Kirchhoff–Helmholtz integral theorem was previously employed to predict the surface pressure and vibrational response of an unbaffled elastic disk to harmonic excitation [J. H. Ginsberg and A. D. Pierce, *J. Acoust. Soc. Am. Suppl.* 1 79, S35 (1986)]. Here, the analysis is extended to describe a finite rigid baffle by using assumed mode functions for the pressure distribution along the vibrating and rigid surfaces. Versions using various analytical functions, as well as finite element modes, are developed and examined for convergence, numerical accuracy, and efficiency. Dependence of the surface

pressure on the baffle radius is examined; results in the limit of an infinite baffle are shown to agree well with a prior analysis by Alper and Magrab [*J. Acoust. Soc. Am.* 48, 681–690 (1970)]. [Work supported by the Office of Naval Research, Code 1132-F.]

8:50

A2. Modal contributions of a finite plate to power spectra. Sung H. Ko (Naval Underwater Systems Center, New London, CT 06320)

The purpose of this work is to compare the modal contribution of a finite elastic plate to the power spectrum and the flexural wave contribution of an infinite elastic plate to the power spectrum. Flexural waves in an infinite plate submerged in a fluid produce pressure waves in the fluid that

travel parallel to the plate and decay away from the plate surface in the fluid. These waves may be called evanescent waves and are strongest in a region near the plate where they may represent noise. When a rectangular plate submerged in a fluid is excited, the vibrating plate generates pressure waves that decay away from the plate surface in the fluid and an acoustic wave that radiates into the fluid. The analytical model considered here is a two-sided fluid-loaded rectangular plate simply supported at its boundaries and excited by a forcing function on one side of the plate. Theoretical analyses are discussed and the wavenumber-dependent transfer functions that are used for calculating power spectra are presented. [Work supported by NUSC.]

9:05

A3. Impulse responses and radiation impedances of cylindrical vibrators using wave-vector-time domain ($k-t$) methods. Peter R. Stepanishen and D. D. Ebenezer (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881-0814)

A wave-vector-time domain ($k-t$) method is presented to evaluate the harmonic and time-dependent loading on a cylindrical shell that is vibrating with a specified spatial and time-dependent velocity. The method is based on utilizing a specified time-dependent modal expansion for the radial velocity of the vibrator. The acoustic loading on the vibrator is also expressed as a modal expansion in which each coefficient is a summation of convolution integrals of the modal velocity coefficients with mode-dependent radiation impulse responses. In contrast to an earlier work [D. D. Ebenezer and Peter R. Stepanishen, *J. Acoust. Soc. Am.* **81**, 854-860 (1987)], the radiation impulse responses are evaluated using a ($k-t$) method based on a time-dependent Green's function for a baffled cylindrical vibrator. A comparison of the new method with the earlier method is presented along with numerical results for various axial mode shapes and circumferential mode numbers. [Work supported by ONR.]

9:20

A4. An array to produce convected normal velocity and pressure fields for wavenumber calibration and boundary layer modification. H. C. Schaub, L. Dwight Luker, and S. Petrie (Naval Research Laboratory, Underwater Sound Reference Detachment, P.O. Box 8337, Orlando, FL 32856-8337)

A transmitting array is described that is capable of producing specified normal velocity distributions at the surface of the array. Such an array is useful in the production of convected normal velocities and pressure fields that may emulate turbulent boundary layer pressure fields. The pressure fields are convective in nature, moving parallel to the plane of the array with a phase velocity that is slow, i.e., less than the speed of sound in the medium. A slow phase velocity corresponds to a spatial wavenumber k that is greater than the corresponding acoustic wavenumber in the medium. Applications include boundary layer modification, testing of sensors in controlled TBL flows, and wavenumber calibration of sensors. Results are presented for several prototype arrays involving different construction methodologies. Shading schemes for the production of uniform convected pressure fields are discussed relative to the physical limitations of the device and electronics. Measured pressure fields and phase velocity are contrasted with those predicted theoretically. Plans for producing larger and more complex arrays are discussed relative to current prototype results. [Work partially supported by ONT.]

9:35

A5. Spherical nearfield calibration array for three-dimensional scanning. A. L. Van Buren (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 8337, Orlando, FL 32856-8337)

Nearfield calibration arrays (NFCAs) are used to determine the far-field acoustic radiation from a transducer by measurements made in the nearfield. The original Trott NFCAs were planar arrays. More recently,

cylindrical NFCAs have been developed. This paper describes a spherical NFCA for determining the full three-dimensional radiation pattern of an enclosed transducer. In theory, the NFCA consists of a large number of discrete hydrophone elements arranged over a spherical surface in N equispaced constant latitude bands. In practice, the array can be synthesized by use of a single semicircular arc containing relatively few elements and by either rotation of the transducer or revolution of the arc about the polar axis. Individual hydrophone sensitivities (both amplitude and phase) are selected so that the NFCA serves as a plane-wave filter for acoustic radiation from inside the sphere. The relative sensitivities, called shading coefficients, are computed using the NFCA reciprocity principle and a least-squares procedure. Only a few sets of $N/2$ shading coefficients are required for full three-dimensional operation of the NFCA over a frequency range of three octaves or more. Design criteria are discussed and sample numerical results are presented.

9:50

A6. Design of an experiment to measure the acoustic mutual impedances of a resonant, close-packed sonar array. Stephen C. Thompson (Acoustics and Sensor Systems, Ocean Systems Division, Gould, Inc., 18901 Euclid Avenue, Cleveland, OH 44117)

Knowledge of the acoustic mutual impedance coefficients for a sonar array is necessary to achieve an accurate analytical description of the performance of the array. In the early stages of a design, analytical approximations, e.g., of an array in an infinite rigid plane baffle, provide adequate estimates. When prototype hardware is available, it may be appropriate to derive refined estimates of the impedance coefficients. This allows more accurate prediction of the effects of design changes on performance, and also provides an analytical baseline for a design that is entering production. The measurement of the mutual impedance coefficients in a resonant, close-packed array is difficult. The only previously reported attempt [Stephen C. Thompson, *J. Acoust. Soc. Am. Suppl.* **1 68**, S34 (1980)] achieved inconclusive results, due to inaccurate estimates of the element electromechanical transfer matrices and strong element interactions that occur in the vicinity of the element resonance frequency. These measurements are being repeated, with greater care given to the previously noted deficiencies. The experiment will be described, concluding with the requirements placed on the special transducer elements that were necessary for the measurement. The design of these transducer elements will be described in a companion paper in this session [M. P. Johnson, *J. Acoust. Soc. Am. Suppl.* **1 82**, S2 (1987)].

10:05

A7. Design of a transducer element for an acoustic mutual impedance measurement experiment. Michael P. Johnson (Acoustics and Sensor Systems, Ocean Systems Division, Gould, Inc., 18901 Euclid Avenue, Cleveland, OH 44117)

A companion paper in this session has described an experiment to measure the acoustic mutual impedances in a resonant, close-packed sonar array [S. C. Thompson, *J. Acoust. Soc. Am. Suppl.* **1 82**, S2 (1987)]. This experiment requires the use of special measurement transducers in place of the elements in the array under study. There were several special requirements for these transducers: (1) high mechanical input impedance at the element radiating face; (2) accurate knowledge of the transducer electromechanical impedance matrix; and (3) radiating face dimensions that are identical to those of the elements of the array under study. The first requirement reduces the effects of interelement coupling and, consequently, reduces the sensitivity in the calculations to errors in the measurement. The second acknowledges that the electromechanical impedance matrices of the transducer elements are needed in the calculation. The third requirement provides a measurement array with the same mutual impedance coefficients as the array under study. A prototype transducer element that seems to meet these requirements is a doubly resonant piston element. The element is designed so that the operating band of the array under study falls between the two resonant frequencies of the measurement transducer. The design and performance of this element will be described.

A8. A diffuse field sonar tank for measurement of projector sound power level and directivity index. David Lubman (D. Lubman & Associates, 14301 Middletown Lane, Westminster, CA 92683)

Though widely used in air acoustics, the diffuse field technique has not been extensively used in underwater acoustics. This paper describes a small (2 m^3) diffuse field sonar tank facility intended for production testing of the steady-state sound-power level and directivity index of wide-band sonar projectors. When evaluated by standards used in air acoustics to ensure adequate diffuse field measurement accuracy, this compact and inexpensive facility appears to be qualified for the determination of band-averaged sound-power level over a frequency range exceeding five octaves (from about 4 to 160 kHz). When compared with a free-field measurement facility, the practical advantages of this approach include large savings in cost and space. Long transducer rise times are permitted since measurements are steady state. Moreover, the implicit spatial integration of radiation patterns makes results of diffuse field testing more appropriate than free-field testing for some applications.

10:35

A9. Practical acoustic beams and synthesized source fields for turbulence detection at the lower atmosphere. S. Adeniyi Adekola (Lagos State University (FETES), Badagry Expressway, Ojo, Lagos, Nigeria)

Computer simulations of new analytical results that are important extensions of an earlier work [S. A. Adekola, *J. Acoust. Soc. Am.* **76**, 345–368 (1984)] are here presented for the echosonde (acoustic echosounding) system. It is shown that the source distributions synthesized from the preassigned directive pressure fields are not only confined within finite antenna aperture regions, but are also characterized by maximum intensities at the boresight region of the aperture and are considerably attenuated towards the rim of the antenna cuff. It is also shown that if the beamwidth of the pattern prescribed is extremely narrow, then the source distribution synthesized from it tends to produce an undesirable strong field or singularity at the rim of the antenna cuff. The paper then focuses attention on

the factors governing the realizations of practical echosonde patterns; which are suitable for turbulence detection of atmospheric irregularities, such as thermal structures, dynamics, and turbulent velocity fields at the lower atmosphere; and from which physically realizable source fields exhibiting no abrupt discontinuities across the antenna aperture can be synthesized. Last, a comparative analysis shows that the approximate patterns generated from the synthesized source distributions manifest good mean-square fits to the idealized acoustic beams originally specified almost all over the entire visible ranges of the patterns prescribed. [Work supported by Lagos State University, Badagry Expressway, Ojo, Lagos, through a Visiting Professorial Appointment.]

10:50

A10. Development of higher-order Zernike polynomials employed in the analysis of acoustic sensing antennas. S. Adeniyi Adekola (Lagos State University (FETES), Badagry Expressway, Ojo, Lagos, Nigeria)

Zernike circle polynomials have been found to be invaluable analytical tools for investigating the characteristics of acoustic sensing antennas [S. A. Adekola, *Acustica* **47**, 114–131 (1981)]. New analytical results obtained from an investigation of the generalized $R_n^m(r)$ Zernike circle polynomials, summarized in the above-mentioned reference, are of higher degrees of accuracy than those presently available in the literature. Indeed, these polynomials are generated to 30° and 29° of freedom describing the even- and odd-numbered modes, respectively; and the field radiated by the acoustic sensing antenna, which is then expressed in terms of the polynomials, is evaluated through direct routine integration. Some of the new analytical results evaluated and numerically computed are here displayed in graphical forms for completeness. Also, illustrative examples given show not only how the Zernike polynomials can be employed in the interpretations of the radiation characteristics of the acoustic echosounding systems, but also how use can be made of the polynomials to describe significantly tapered source distributions essential in successful acoustic remote sensing of the lower atmosphere. [Work supported by Lagos State University, Badagry Expressway, Ojo, Lagos, through a Visiting Professorial Appointment.]

TUESDAY MORNING, 17 NOVEMBER 1987

UNIVERSITY LECTURE HALL, 8:30 A.M. TO 12:00 NOON

Session B. Psychological and Physiological Acoustics I: Hearing-Impaired Speech and Auditory Perception

Joseph W. Hall, III, Chairman

Division of Otolaryngology, University of North Carolina, 610 Burnett-Womack Building, 229H, Chapel Hill, North Carolina 27514

Contributed Papers

8:30

B1. The effect of varying the amplitude-frequency response on the masked SRT for sentences in hearing-impaired listeners. Janette N. van Dijkhuizen, Joost M. Festen, and Reinier Plomp (Experimental Audiology, ENT Department, Free University Hospital, Amsterdam, The Netherlands)

A multichannel gain-control hearing aid, in which the frequency-dependent amplification is adapted automatically to the fluctuations of the incoming signal, may optimally deliver speech to an impaired ear. Such a system requires that the speech-reception threshold (SRT) in noise is,

within limits, unaffected by dynamic variations in the amplitude-frequency response of the hearing aid. For normal-hearing listeners, van Dijkhuizen *et al.* [*J. Acoust. Soc. Am.* **81**, 465–496 (1987)] found that the masked SRT for sentences is remarkably resistant to dynamic variations in the slope of the amplitude-frequency response when it shaped both speech and noise. In this experiment, we studied corresponding conditions for hearing-impaired listeners. Again, the amplitude-frequency response shaped both speech and noise and thus left speech-to-noise ratios untouched. The results obtained so far also indicate that, for hearing-impaired listeners, dynamic variations of the amplitude-frequency response do not affect the SRT.

B2. Explorations on the difference in SRT between a stationary noise masker and an interfering speaker. Joost M. Festen (Experimental Audiology, ENT Department, Free University Hospital, Amsterdam, The Netherlands)

For moderate hearing impairment, the speech-reception threshold (SRT) for sentences in stationary noise is roughly up to 5 dB higher than for normal hearing. However, a fluctuating interfacing sound seems more typical for everyday listening conditions than stationary noise. For a group of young hearing-impaired listeners, the difference in SRT with normal-hearing listeners increased from 2.7 dB for stationary noise to 7 dB for 100% sinusoidally intensity-modulated noise, and even to 10.5 dB for interfering speech. In this experiment, the interfering speech was the time-reversed voice of the same speaker as the signal, which may have introduced an extra difficulty. In a subsequent experiment, the SRT for sentences from two speakers (one male and one female) was measured with a variety of fluctuating maskers covering the range from stationary noise to interfering speech. The two voices were mutually used as masker, giving a meaningful interference. Meaningless interference was obtained by time reversal of the interfering speech or by modulating noise with the envelope of speech. Two types of modulation were used: one broadband and one in which high- and low-frequency noise (separation frequency 1000 Hz) was modulated with the high- and low-frequency envelope of speech, respectively.

9:00

B3. Reduced auditory discrimination and compression in hearing aids. Wouter A. Dreschler (Academic Medical Center, Clinical Audiology, University of Amsterdam, Amsterdam, The Netherlands)

Two specific applications of single-channel compression are studied separately: whole-range compression as a compensation for recruitment, and compression limiting as an elegant method to limit excessive output levels. In cases of compression limiting, compression proved to be superior to peak clipping, especially for the perception of consonants: An average improvement in identification scores of 18% was found and less roll-over for the perception of final consonants at high presentation levels. In cases of whole-range compression, a small, but significant, increase in identification scores was found for settings with the compression knee-point at the lowest level. A detailed analysis of the patterns of confusions revealed clear qualitative differences in the perception of phonemes with and without compression. These differences can be brought into relation with an improved perception of temporal cues, like the preburst silent interval of plosives, and with spectral changes due to the activation of the compression circuit.

9:15

B4. Correlations among auditory functions in the "normal" range. Mark Haggard, Gabrielle Saunders, and Stuart Gatehouse (MRC Institute of Hearing Research, Nottingham NG7 2RD, United Kingdom)

To date, hearing levels and psychoacoustic measures have shown factor structure and correlations with speech perception in noise among the hearing impaired, but not among clinically "normal" ears. A pseudofree-field sentence-in-noise test (PFFIN) from a dummy-head recording, simplified psychoacoustic tuning curves, and tests of linguistic processing were administered to 60 people with hearing better than 20 dB HL (0.5–2.0 kHz); one-third had complained of auditory difficulties. Significant

correlations were found among the psychoacoustic variables, which also predicted PFFIN score. Correlations between hearing levels and PFFIN remained significant after partialing age, noise exposure, upward spread of masking at 2 kHz, and educational and linguistic measures. Systematically truncating the HL distribution showed that this correlation lies mostly in the 15–25 dB HL range and implicates low- as well as high-frequency hearing. Thus the 17% of the adult population between 15 and 25 dB HL (0.5–4.0 kHz) should be considered marginal, both clinically and experimentally. Scientifically, if not pragmatically, a "low fence" at 15 dB HL is justified.

9:30

B5. Evaluating the Articulation Index for auditory-visual input. Ken W. Grant (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

The current standard for calculating the Articulation Index (AI) includes a procedure to estimate the effective AI when hearing is combined with speechreading [ANSI S3.5-1969 (R1978), "Methods for the Calculation of the Articulation Index" (American National Standards Institute, New York, 1969)]. This procedure assumes that the band-importance function derived for auditory listening situations applies equally well to auditory-visual situations. Recent studies have shown, however, that certain auditory signals that, by themselves, produce negligible speech reception scores (e.g., F_0 , speech-modulated noise, etc.) can provide substantial benefits to speechreading. The existence of such signals suggests that the band-importance function for auditory and auditory-visual inputs may be different. In the present study, an attempt was made to validate the auditory-visual correction procedure outlined in the ANSI-1969 standard by evaluating auditory, visual, and auditory-visual sentence identification performance of normal-hearing subjects for both wideband speech degraded by additive noise and bandpass-filtered speech presented in quiet. The results obtained for auditory listening conditions with an AI greater than 0.03 support the procedure outlined in the current ANSI standard. [Work supported by NIH.]

9:45

B6. Frequency discrimination and consonant recognition in the hearing impaired. Marleen T. Ochs (Department of Communication Sciences and Disorders, Radford University, Radford, VA 24142)

The frequency difference limen (DL) for short-duration second-formant transitions (F_2) presented alone and in speechlike environments was examined in normal hearing and hearing-impaired subjects. Four stimulus conditions were included: F_2 transition alone; F_2 transition in the presence of all formants (full formant); full-formant stimuli preceded by a burst and followed by a vowel; and full-formant stimuli preceded and followed by a vowel. For the simplest stimulus condition (F_2 alone), all hearing-impaired subjects had DLs within the 95% confidence interval around the mean of the normals. For all other conditions, however, the hearing impaired demonstrated extreme intersubject variation. Some hearing-impaired subjects could not detect a formant transition of even 800 Hz, whereas others continued to perform like normal hearers. Consonant identification performance was also examined in CV and VCV environments. Consonants whose identification is believed to involve formant transitions were selected for study. In spite of modifications to the frequency discrimination task and use of relevant syllables, relatively weak correlations continue to be observed between F_2 DL and consonant recognition ability, as well as between F_2 DL and degree of hearing loss. [Work supported by NIH.]

10:00–10:15

Break

B7. Lateral suppression and auditory frequency selectivity. Wouter A. Dreschler and A. Rens Leeuw (Academic Medical Center, Clinical Audiology, University of Amsterdam, Amsterdam, The Netherlands)

For measuring frequency selectivity, the notched-noise procedure, introduced by Patterson and Nimmo-Smith, is regarded at the moment as the best method to avoid cueing effects and to take off-frequency listening into account. In this study, a similar approach was adopted for the measurement of lateral suppression. The amount of suppression for a two-tone masker, as a function of the frequency of the suppressor, was measured in the presence of a notched noise. Also in this paradigm, a fair amount of suppression was found in a group of 15 normal-hearing subjects. For hearing-impaired subjects, and, surprisingly, also for subjects with a conductive loss, a gradual loss of suppression as a function of hearing loss was observed. Finally, the results are related to auditory filter shapes obtained in simultaneous and forward masking.

10:30

B8. Frequency masking patterns and "excess masking" in normal-hearing and hearing-impaired listeners. Margery A. Garrison, Anna C. Schroder, and David A. Nelson (Hearing Research Laboratory, University of Minnesota, 2630 University Avenue S.E., Minneapolis, MN 55414)

The presence of "excessive upward spread of masking" in high-frequency sensorineural hearing-loss (HFSNHL) subjects has been previously demonstrated by other investigators. This excess masking, defined by predictions from a "noise" model of hearing loss, occurs in the transition region between normal and impaired hearing. To examine this phenomenon further, frequency masking patterns (FMPs) were obtained from two normal-hearing subjects and one HFSNHL subject. Using a 200-Hz narrow-band noise (NBN) masker (90 dB SPL) with an upper edge at 520 Hz, FMPs from the impaired ear demonstrated excess masking, which it was hypothesized might be explained by an inability to "listen in the valleys" of the masker envelope because of poor temporal resolution. To test this notion, the experiment was repeated using a 40-Hz NBN masker (90 dB SPL), which has less rapid envelope fluctuations. No differences in the upward spread of masking were seen between the 200- and 40-Hz bandwidth-equal SPL maskers. When the experiment was repeated with equal spectrum level maskers, the 40-Hz NBN masker produced less upward spread of masking in both normal-hearing and hearing-impaired ears. These data suggest that, for the HFSNHL subject, excessive upward spread of masking cannot be completely explained by an inability to "listen in the valleys." [Work supported by NINCDS.]

10:45

B9. High-frequency audiometry in platinum ototoxicity. Wouter A. Dreschler and Rob J. A. M. v/d Hulst (Academic Medical Center, Clinical Audiology, University of Amsterdam, Amsterdam, The Netherlands)

In a group of 144 subjects (288 ears) medically treated with platinum drugs, high-frequency audiometry was applied at frequent intervals in order to compare the effects of three different drug administrations: high-dose *cis*-platinum cures, low-dose *cis*-platinum cures, and carboplatin cures. In all subgroups, high-frequency audiometry considerably enhanced the early detection of ototoxicity. Marked differences between cures were established, both in the pattern of onset of the damage and in the relation between dose and damage severity. For subjects treated with platinum derivatives, especially, the thresholds at 14 kHz prove to be important. The results suggest that, for these subjects, a single measuring frequency should be taken into consideration. With a minimum of effort, most of the increased sensitivity for a complete high-frequency audiogram can be reached. Finally, the predictive value of a threshold deterioration above 8 kHz for a threshold deterioration in the conventional range of audiometric frequencies will be considered.

B10. Sinusoidal modeling of speech: Its use in simulating and compensating for recruitment of loudness. Janet R. Lias and Mark A. Clements (School of Electrical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Amplitude compression techniques utilizing both single and multi-channel filtering systems have been proposed for use in compensating for recruitment of loudness in hearing aids. Due to a variety of limitations, these techniques have shown only marginal success. The sinusoidal model of speech developed by Quatieri and McAulay [IEEE Trans. Acoust. Speech Signal Process. ASSP-34, 744-754 (1986)] has been used to simulate and compensate for recruitment, using the general theory described by Villchur [J. Acoust. Soc. Am. 56, 1601-1611 (1974)] and others. Specifically, the model synthesizes speech as the sum of sinusoids with time-varying amplitudes and phases. Amplitude compression and amplification can be performed on individual sinusoids to shape speech to fit an impaired person's residual hearing. Conversely, expansion can be used to simulate the effects of recruitment. This model offers almost unlimited flexibility in areas such as consonant boosting and spectral shaping in any portion of the spectrum. With a real-time implementation of this model, the parameters can be easily adjusted to suit each person's needs.

11:15

B11. A simple procedure for measuring iso-loudness contours. J. B. Allen (AT&T Bell Laboratories, Murray Hill, NJ 07974), P. S. Chien (Center for Research in Speech and Hearing Sciences, City University of New York), and J. L. Hall (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A procedure for measuring iso-loudness contours in normal and hearing-impaired subjects similar to that used by Geller and Margolis has been devised [J. Speech Hear. Res. 27, 20-27 (1984)]. The stimuli are bursts of noise $\frac{1}{2}$ -oct wide, centered at 1-oct intervals from 250 Hz to 4 kHz. The subject is asked to rate the stimuli according to a six-point scale, using the descriptive words "very soft," "soft," "OK," "loud," "very loud," and "too loud." The instructions include a one-sentence description of each loudness category. Failure to respond is recorded as "nothing." In an initial pass, the lower and upper limits of the subject's hearing range at each frequency are estimated. These initial trials also serve to train the subject. Following the initial pass, 15 levels, logarithmically spaced between the measured limits at each frequency, are presented in random order, three times each. The entire procedure takes about 30 min to complete, and it provides stable estimates (based on test-retest measurements) of loudness growth functions. These loudness growth functions are then replotted as iso-loudness contours. Data from normal and hearing-impaired subjects will be presented along with their pure-tone thresholds.

11:30

B12. Effects of fluctuating conductive hearing loss in infancy on auditory brain-stem responses and binaural interaction measured at age 5. Adele Gunnarson and Terese Finitzo (University of Texas at Dallas/Callier Center, 1966 Inwood Road, Dallas, TX 75235)

The purpose of this research was to examine long-term auditory electrophysiologic effects of early conductive hearing loss. Animal deprivation studies suggest brain-stem anatomic and physiologic alteration during critical periods. Comparable human research is limited. The research questioned whether auditory brain-stem responses (ABR) and binaural interaction (BI) at ages 5-7 differed depending on documented patterns of early hearing. Stimuli were rarefaction clicks at 21.1/s, presented binaurally at 60 dB and monaurally at 30, 60, and 80 dB HL. The BI was derived by subtracting summed ipsilateral and contralateral monaural responses from binaural responses, and defined as deflections whose amplitude exceeded 3 s.d. from derived prestimulus amplitudes. Results indicated that waves III and V, and I-III and I-V interpeak latencies (IPL) differed when controls were compared to experimental subjects. Wave I, the latency/intensity functions at 30-60 and 30-80 dB, and III-V IPLs

were unchanged. The BI was present in 8 out of 9 controls and 8 out of 9 children with fluctuating mild losses in infancy. Only 4 out of 9 children with moderate asymmetric losses in infancy had BI at age 5. [Work supported by NIH No. 5 RO1 NS 19675-03.]

11:45

B13. Cue salience in the perception of a stop voicing contrast by hearing and hearing-impaired children. Valerie Hazan (Department of Phonetics and Linguistics, University College London, 4, Stephenson Way, London NW1 2HE, United Kingdom), Lisa Holden-Pitt, Sally Revoile, Donna Edward, and Janet Droge (Center for Auditory and Speech Sciences, Gallaudet University, Washington, DC 20002)

The contribution of two acoustic cues, voice onset time (VOT) and vowel onset frequencies, to the perception of a TAD-DAD voicing contrast was examined for ten normal-hearing and six hearing-impaired children (3 FA: 50-95 dB HL) between 6 and 12 years of age. Both natural stimuli and high-quality synthesized copies of these stimuli were used to create five VOT continua (three natural and two synthetic), in which the vowel onset cue was either appropriate or conflicting. An adaptive identification procedure was used to determine phoneme boundaries under these five test conditions. Changes in spectral characteristics at vowel onset did produce significant shifts in phoneme boundary for the normal-hearing group, but not for the hearing-impaired group. Similar phoneme boundaries were obtained by the normal-hearing group for natural continua and their synthetic counterparts. However, higher values of standard error were observed for synthetic stimuli for both listener groups.

TUESDAY MORNING, 17 NOVEMBER 1987

UMS 4 AND 5, 8:30 TO 11:55 A.M.

Session C. Underwater Acoustics I: Arctic Acoustics I

Dan J. Ramsdale, Chairman

Naval Ocean Research and Development Center, NSTL Station, Mississippi 39529

Chairman's Introduction—8:30

Invited Papers

8:35

C1. Geoacoustics of the shallow-water Arctic. J. E. Matthews (Numerical Modeling Division, Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

From the perspective of geology, the shallow-water regions of the Arctic Ocean can be divided into two provinces of approximately equal size. One is characterized by unglaciated, thick, prograding sedimentary deposits, which are found on the continental shelves of the Beaufort, Chukchi, and East Siberian Seas. The other is characterized by a glaciated bank-trough terrain, and includes the continental shelf of the Canadian-Greenland Arctic and the Barents, Kara, and Laptev Seas. From a low-frequency geoacoustics point of view, the prograding shelf province consists of a few to tens of meters of poorly sorted clastic sediment (1.5-1.6 km/s) that overlies an acoustic basement of consolidated coarse sediment (1.7 km/s). The bank-trough province has two distinctly different environments: shallow bands that consist of zero to several meters of coarse sediment (1.6-1.7 km/s) overlying an acoustic basement of sedimentary rock (3.5 km/s), and deep troughs floored by a few tens of meters of poorly sorted sediment (1.5-1.6 km/s) over an acoustic basement of consolidated sediment (1.9 km/s). These generalizations must be considered tentative because of the critical scarcity of data; however, plane-wave reflection and normal-mode calculations are presented for these generic environments to demonstrate their acoustic similarities and differences.

9:00

C2. Sound-speed structure of the Arctic Ocean and the Greenland Sea Marginal Ice Zone. John L. Newton (Polar Research Laboratory, Carpinteria, CA 93013)

Within the central ice-covered Arctic Basin, well away from marginal influences, the vertical sound-speed profile can be characterized by four positive sound-speed gradient layers. These layers (and the approximate gradients) are a surface mixed layer extending to a depth of a few tens of meters ($\sim 0.016/s$); a layer from the base of the mixed layer to the depth (200-500 m) of the temperature maximum in the Atlantic water ($\sim 0.080/s$); a layer from the temperature maximum to the bottom of the Atlantic water at about 1000 m ($\sim 0.010/s$); and the deep water extending to the ocean bottom ($\sim 0.016/s$). The consistent and notable differences evident in the general sound-speed structure of the Canadian and Eurasian Basins can be related to the large scale Arctic Ocean circulation. Profile-to-profile variability and the occurrence of small vertical scale sound-speed features, such as relative minima/maxima and steplike structures, are significantly higher in the Eurasian Basin than in the Canadian Basin. The sound-speed distribution within the Greenland Sea Marginal Ice Zone is dominated by the presence of a strong oceanographic front, the East Greenland Polar Front, which marks the boundary between the cool, fresh Polar water exiting the Arctic Ocean and the adjoining warmer, more saline Atlantic water. Because of this strong ocean front, profile-to-profile sound-speed variability in the Marginal Ice

Zone is high over short distances and times. The complex sound-speed structure and its high variability result from various interactions between the two water masses which can take the form of eddies, intrusive layers, and steplike features.

9:25

C3. Recent advances in Arctic high-frequency acoustics. R. E. François (Applied Physics Laboratory, University of Washington, Seattle, WA 98195)

In the past few years, research in Arctic high-frequency acoustics ($f > 5$ kHz) has brought about a much better understanding of the effect of environmental factors on acoustics. Our knowledge of the effect of water properties continues to increase as well-known phenomena are routinely treated in greater detail. Ice properties, on the other hand, and the effect of these properties on acoustics, require new ideas and diligent research. For example, and perhaps of foremost importance, consider reflections from sea ice. Studies of the reflecting properties of the lower surface of sea ice, with its dendritic layer, show that at low frequencies the reflection coefficient at normal incidence depends only on the bulk properties of the medium. At higher frequencies (up to 200 kHz) Gaussian distributions for reflection coefficients are observed, with standard deviations that increase with frequency and averages that decrease with frequency, as much as 28 dB below that predicted by bulk properties alone. Studies of the target strength characteristics of ice block shapes have shown that rigid body theory for simple shapes, while good for hard ice surfaces, only crudely predicts actual response for freezing surfaces. As for *in situ* ice keels, the variation in return with frequency shows that reflections are often the coherent sum of several reflectors near the same range and close enough in azimuth that such returns appear to be pointlike. [Work is supported by Office of Naval Technology.]

9:50

C4. A review of applied Arctic acoustic models. R. C. Cavanagh, J. A. Davis (Planning Systems, Inc., 7925 Westpark Drive, McLean, VA 22102), and J. S. Hanna (SAIC, 1710 Goodridge Drive, McLean, VA 22102)

Basic acoustic model development related to the Arctic environment has been carried out by various researchers over the past 25 years. This work is being incorporated into applied models for propagation and noise that attempt to provide useful predictive capability given suitable environmental data bases. A review is presented of the candidate theories and some history of their development. In particular, ice scattering theories are described that show promise of utility and their incorporation into the several standard propagation models is discussed. Some comparisons of these models with data are presented and the outstanding issues related to model development and the adequacy of existing data are presented. Although the development of a corresponding ambient noise model is a much less mature activity, current attempts at such development are reviewed. [Work supported by the ASW Environmental Acoustic Support Program.]

10:15

C5. High-frequency acoustic backscatter from the Arctic ice canopy: Theory and experiment. Suzanne T. McDaniel (Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, PA 16804)

The method of physical optics is applied to treat the backscatter of high-frequency acoustic signals from the Arctic ice canopy. Theoretical results are obtained for both the mean backscattering strength and the spatial coherence of the scattered field. The mean backscattering strength is found to be dependent on the fractional surface area covered by underwater ice ridges. Using measured ice distributions, the variability in mean scattering strength is predicted to vary by at most ± 2.5 dB from its average value for a wide variety of ice conditions. The theory is then applied to treat the spatial coherence of backscatter. The coherence is found to depend on the ice draft distribution and ensonified surface area. The results of experiments to measure the vertical spatial coherence of bistatic backscatter are presented. These results not only confirm the theoretically predicted dependences on measurement geometry but also support predictions of the relative importance of ice roughness versus ensonification. [Research supported by ONT with technical management by NORDA.]

10:40

C6. Energy partitioning for very-low-frequency, long-range acoustic propagation in the Arctic. A. B. Baggeroer (Massachusetts Institute of Technology, Cambridge, MA 02139), G. L. Duckworth (Bolt Beranek and Newman Laboratories, Cambridge, MA 02238), and E. K. Scheer (Woods Hole Oceanographic Institute, Woods Hole, MA 02543)

In long-range acoustic propagation in the Arctic, very-low-frequency signals interact with both the ice canopy and the bottom, both of which introduce loss. Signals from a sequence of long-range (340-km) shots detonated at several depths were recorded on a multichannel array and resolved according to their horizontal phase velocity and frequency components to separate the loss effects of these two boundaries. Source levels and geometric losses based upon a determined sound-speed model were subtracted to arrive at an absorptive/scattered loss measure for each resolved path and frequency. These reduced data were used to estimate the

partitioning of energy according to phase velocity and frequency from the ice canopy and the seafloor. At frequencies below 20 Hz, bottom interacting signals account for more than 50% of the received energy observed at the array on almost all the shots.

11:05

C7. Arctic ambient noise mechanisms. Ira Dyer (Massachusetts Institute of Technology, Department of Ocean Engineering, Cambridge, MA 02139)

The study of ambient noise can be divided into two categories, one dealing with the correlation of averaged noise with the slow evolution of environmental parameters, and the other concentrating on details of the ice fracture processes that ultimately give rise to individual noise events. Recent progress in both categories, especially at low frequencies (3–100 Hz) where significant results appear to be at hand, is reviewed. Thoughts on higher frequencies (100–3000 Hz) are also given, for which it seems likely that the environmental correlates and detailed source mechanisms are of a different character than at low frequencies. Finally, distinctions are drawn between pack ice of the central Arctic and low-concentration ice of the marginal ice zone, with respect to their likely mechanisms of noise generation. [Work supported by ONR, Arctic Program Office.]

11:30

C8. Arctic data buoys—An update. Beaumont M. Buck (Polar Research Laboratory, Inc., 6309 Carpinteria Avenue, Carpinteria, CA 93013)

A 1983 paper, "Arctic data collection the easy way—Data buoys" [J. Acoust. Soc. Am. Suppl. 1 74, S2 (1983)], presented data buoys as a partial solution to the problems and limitations inherent to manned operations in the Arctic. The present paper describes the successes and failures of these devices, as applied to Arctic underwater acoustics, in the intervening years and what has been done with the large quantities of data that they have generated. This will include wide-area, synoptic collection of ambient noise level data, as well as first-ever measurement of low-frequency propagation during the summertime in the Central Arctic using autonomous cw projector and receiving data buoys. Plans for even more sophisticated devices to be installed in 1988 are discussed. [Work supported by U.S. Navy.]

TUESDAY MORNING, 17 NOVEMBER 1987

POINCIANA ROOM, 9:00 TO 10:40 A.M.

Session D. Architectural Acoustics I: Acoustical Design of Renovated Performance Facilities I

Christopher Jaffe, Chairman

Jaffe Acoustics, Inc., 114A Washington Street, Norwalk, Connecticut 06854

Chairman's Introduction—9:00

Invited Papers

9:10

D1. The movie palace as performing arts center. Peter J. George (Peter George Associates, Inc., 34 West 17th Street, New York, NY 10011)

Many movie theaters in downtown neighborhoods have been saved from demolition by their renovation and recycling as performing arts centers. While these spaces often have physical characteristics that are deemed acoustically desirable, there are often many features that are detrimental to a good acoustical performing environment. This paper describes typical attributes and deficiencies with methodology to improve the latter.

9:40

D2. Some unusual renovation projects. David L. Klepper (Klepper Marshall King Associates, Ltd., 7 Holland Avenue, White Plains, NY 10603)

A comparison of several different renovation projects indicates that an acoustical consultant must be very flexible in adopting a particular set of recommendations for the acoustical, architectural, and restoration

requirements of a particular project. In one project, sound-reflecting panels may be appropriate, in another, definitely not. In one project, an ideal central cluster loudspeaker system may be appropriate, in a second, it must be supplemented by delayed underbalcony loudspeakers, while a third would best employ vertical line-source loudspeakers, a fourth a horizontal line source, and a fifth a distributed delayed loudspeaker system. By means of slides, the differences between these several projects, which include a university concert hall, the conversion of a motion picture theatre to a multi-use performing arts center with emphasis on symphony concerts, a university theatre, and two 19th century opera houses restored as multi-use performing arts centers will be explored. One common thread runs through all of them, however; all make maximum use of live sound before the use of electronics, with minimum use of sound-absorbing treatment.

10:10

D3. Adaptive reuse of existing auditoria. J. Christopher Jaffe (Jaffe Acoustics, Inc., 114A Washington Street, Norwalk, CT 06854)

The adaptive reuse of existing auditoria is an extremely cost efficient way for communities to create contemporary "state of the art" performance facilities. The key to the successful culmination of such a facility is a clear definition of the program use of the space and an understanding between the client and the consultant that all available acoustic techniques can be utilized to achieve final design goals. This paper will discuss the adaptive reuse of a number of buildings as diverse as a library, a union hall, a WPA built art deco music hall, and a three theatre/film complex.

TUESDAY MORNING, 17 NOVEMBER 1987

TUTTLE SOUTH ROOM, 9:00 TO 11:20 A.M.

Session E. Noise I: Properties and New Applications of Porous Material

Gilles A. Daigle, Chairman

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

Chairman's Introduction—9:00

Invited Papers

9:05

E1. Theories for sound propagation in rigid-framed porous materials. Keith Attenborough (Engineering Mechanics Discipline, Faculty of Technology, The Open University, Milton Keynes MK7 6AA, United Kingdom)

An historical overview of the development of theories for sound propagation in rigid-framed porous media is offered. Phenomenological and microstructural theories are contrasted. Microstructural approaches may start either from specifications of the geometrical form of the solid frame or from specifications of the geometrical form of the pores. The former approach is particularly useful for describing acoustical properties of fibrous materials. The latter approach has the longest history, the wider application, and is the basis of many recent developments in the dynamic theory of poroelasticity. In one form of the pore-based theory, consideration is given to the incorporation of dynamic effects of pore shape and irregularity and steady flow tortuosity. An alternative form results from a more rigorous approach to the effects on viscous drag and thermal interaction with pore walls of the inclination of pore axes to the macroscopic pressure gradient, i.e., dynamic tortuosity effects. By comparing theoretical predictions with experimental data for sand, lead shot, glass beads, and snow, both the usefulness and the current limitations of theories that use microscopic pore-based descriptions and their various approximations are explored.

9:30

E2. Towards a method for predicting absorption coefficient rather than measuring it in a reverberation chamber. J. Nicolas (G.A.U.S., Génie mécanique, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

The modelization of the impedance of fibrous and rigid process material is discussed. For fibrous soft material such as fiberglass, it will be shown that Delany and Bazley's model [Appl. Acoust. 3, 105-116

(1970)] is quite reliable for normal specific impedance as well as for oblique incidence. Experimental validations are done via the two-microphone method [submitted (1987)] (normal and oblique) and via the indirect method of the equivalent flow resistivity (oblique only). Performance of less classical types of soft material like felt is not well quantified by Delany and Bazley, showing clearly the limitation of this one-parameter model. For rigid porous material, the impedance behavior is generally more complex. Products like ceramics and cement-wood fiber have been under investigation. Flow resistivity, porosity, and tortuosity have been measured and introduced in Attenborough's four-parameter model [J. Acoust. Soc. Am. 73, 785-799 (1983)]. This leaves a not easily measurable parameter, the shape factor, which is used as an adjustable parameter. For these rigid materials, good agreement will be found between prediction and experimental data even for the important oscillations encountered for high frequency. This indicates that, with simple setups for the measurement of physical parameter, one may be able to calculate absorption coefficient rather than measuring it especially for normal incidence. However, the link between a calculated diffuse field absorption coefficient and the one measured in a reverberation chamber is not clear at present.

9:55

E3. Optimal use of noise control foams. J. Stuart Bolton (Ray W. Herrick Laboratories, Purdue University, West Lafayette, IN 47907)

Polyurethane foam is used with increasing frequency in noise control applications owing to its ease of handling and manufacturability. Until recently it has not been possible to predict its acoustical behavior with the accuracy that is possible for fiberglass, thus discouraging the use of foam in critical applications. Now theoretical models governing one-dimensional wave propagation in noise control foams have become available that have been shown to be capable of reproducing normal incidence absorption and transmission measurements accurately. These theories have been used to predict the behavior of foam in several absorption and transmission applications. In confirmation of largely anecdotal evidence, it has been found that the acoustical behavior of foam is very sensitive to its method of application to backing surfaces and to the way surface coatings are applied. In addition, it has been possible to identify optimum treatment configurations. These studies will be summarized here. Progress towards full three-dimensional foam models will also be described. Such models are necessary to allow the prediction of oblique incidence sound transmission and absorption. Finally, theory and experiments related to a new use of foam as a vibration damping material will be reported.

10:20-10:25

Break

10:25

E4. Measurement of acoustic impedance of sound absorbing materials in a free field at low frequencies. Y. Champoux and J. F. Allard (Mechanical Engineering Department, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

The two-microphone technique allows the precise measurement of acoustic velocity and consequently the determination of acoustic intensity. Using this principle, a method has been proposed to measure, in a free field, the acoustic impedance of sound absorbing materials. Good results were obtained for frequencies ranging from 700 Hz to 4 kHz [J. F. Allard *et al.*, J. Sound Vib. 114(2) (1987)]. Recently, it has been demonstrated theoretically [Y. Champoux and A.L'Espérance, J. Acoust. Soc. Am. Suppl. 1 80, S56 (1986)] that for low frequencies, even if the source is several meters from the sample, the spherical nature of the waves has to be taken into account. This paper presents experimental evidence that confirms that the modeling correctly describes the phenomena. Further several refinements to the experimental technique, allowing precise measurements for frequencies as low as 250 Hz, will be demonstrated.

Contributed Papers

10:50

E5. The replacement of mineral wool in perforated ceilings by a laminated textile. Glenn H. Frommer (DAMPA A/S, 5690 Tommerup, Denmark)

A layer of 0.2-mm textile laminated onto a perforated metal band has been developed to replace inlays of 12-mm 34-k/m³ mineral wool in

DAMPA ceiling products. Sound absorption values for the textile are 15%-20% better at frequencies under 315 Hz, and generally higher at all other frequencies, NRC = 0.75. This exceptional property is achieved by optimizing the streaming resistance of the textile and its lamination. The ceiling units conform with the requirements of the British Building Regulations, 1976, for a class-0 surface and the ASTM requirements for smoke density. In contrast with other porous materials, vibrational tests show no

evidence of fallout. The Danish Environmental Prize has been awarded for the development of these products.

11:05

E6. The measurement of absorption coefficient and acoustic impedance using spherical waves and a transfer function method. Matthew A. Nobile (IBM Acoustics Laboratory, Department C18, Building 704, P. O. Box 390, Poughkeepsie, NY 12602)

The normal-incidence absorption coefficient (or complex impedance) of an acoustical material is usually measured in a standing wave tube using

plane wave excitation. In order to obtain valid high-frequency data, a tube with a small cross section is required. But, quite often, the absorptive properties of the small sample of material that is tested are not representative of those of the larger sample from which it has been taken. An engineering method has been described [M. A. Nobile, Proc. NOISE-CON '87, Penn State University, 8-10 June 1987, pp. 611-616] that shows that large samples of materials (or outdoor ground surfaces) can be evaluated using spherical waves in a hemi-anechoic environment. As with the standing-wave tube method (per ASTM E-1050), this new method utilizes random noise excitation and computes the transfer function between two closely spaced microphones. The accuracy is limited primarily by diffraction from the edges of the finite sample, which is not taken into account. This paper will address the theoretical aspects of the problem and present some recent experimental results.

TUESDAY MORNING, 17 NOVEMBER 1987 TUTTLE CENTER ROOM, 9:00 A.M. TO 12:05 P.M.

Session F. Physical Acoustics I: Nonlinear Acoustics

Mark F. Hamilton, Chairman

Department of Mechanical Engineering, University of Texas at Austin, Austin, Texas 78712-1063

Chairman's Introduction—9:00

Contributed Papers

9:05

F1. Nonlinear compressibility corrections to induced hydrodynamic mass: Consequences for small amplitude supersonic modes. S. Putterman, P. H. Roberts (Physics and Mathematics Department, University of California, Los Angeles, CA 90024), and W. Fiszdon (Department of Fluid Mechanics, Polish Academy of Sciences, 00-049 Warsaw, Poland)

The induced mass of an object accelerating through an isotropic fluid has contributions due to the fluid's compressibility that are anisotropic and nonlinear in the velocity. This effect can lead to pronounced amplitude dependence in the resonant frequency of supersonic modes. One such example to be presented involves the plasma wave that propagates in 2-D arrays of ions in He⁴ with a velocity that is ten times the He⁴ sound velocity. [Work of S. P. supported by DOE and work of P.H.R. supported by ONR.]

9:20

F2. Sound generation from a transonically moving laser. Rao J. Chandrasekhar and Ilene J. Busch-Vishniac (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78712)

Experimental investigations of the sound generated underwater by a moving high-power laser source have shown roughly a 25-dB boost in the pressure if the source moves at Mach one. A three-pronged approach to studying this transonic amplification phenomenon has been used: an analytical examination of the effect of including first-order entropy production terms in the equation of state, a numerical investigation of the effect of source velocity on the sound generating efficiency, and an experimental test of the validity of superposition as a function of source velocity. Results show that nonlinearities and entropy production cannot explain the amplification phenomenon. Further, the acoustic efficiency is shown to be substantially lower for a transonic source than for a subsonic source. Hence, the transonic amplification phenomenon is still not explainable. [Research supported by ONR.]

9:35

F3. Nonlinearity parameters for superfluid helium. M. S. Cramer and R. Sen (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

The nonlinear propagation of second-sound in He II near one of the zeros of the quadratic steepening parameter associated with the B/A ratio is considered. It is shown that the correct description at these pressures and temperatures requires an additional nonlinearity parameter. New expressions for the nonlinear sound and shock speed are given and compared to experiment.

9:50

F4. Refraction of finite amplitude acoustic waves at a plane fluid-fluid interface. Frederick D. Cotaras and David T. Blackstock (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

One of the oldest and most venerable laws of acoustics is Snell's law. But does this law hold for finite amplitude sound? By following the path of a wavelet (point on a waveform), the following form of Snell's law for a finite amplitude wave is derived:

$$\sin \theta_t / \sin \theta_i = (c_2 + \beta_2 u_t) / (c_1 + \beta_1 u_i),$$

where θ , c , β , and u are angle, small-signal sound speed, nonlinearity coefficient, and particle velocity, respectively. Subscripts i , t , 1, and 2 denote incident wave, transmitted wave, fluid 1, and fluid 2, respectively. Since θ_t is predicted to vary from wavelet to wavelet, the transmitted wave field cannot be truly planar. However, by following the path of individual wavelets, the waveform of the transmitted signal is computed at any point in the field. [Work supported by ONR.]

10:05

F5. Effects of oscillatory motion on crystal growth. Charles Thompson and Vineet Mehta (Department of Electrical Engineering, University of Lowell, 1 University Avenue, Lowell, MA 01854)

This work explores how thermally induced instabilities driven by the oscillatory motion of a heated fluid affect the growth of a semiconductor crystal. The mechanisms governing interaction between the interface temperature and its growth will be presented. It will be shown that the dynamic characteristic growth front can be described by a nonlinear differential equation. Preliminary results of our analysis suggest that morphological defects in bulk materials can be attributed to the accelerated growth rate that occurs in response to high unsteady temperature gradients. The theory formulated suggests methods that could be used in controlling quality of semiconductor crystals. Finally, the implications of this theory in formulating better processes for manufacturing bulk semiconductor materials will be outlined. [Work supported by Analog Devices Professorship.]

10:20

F6. Effects of boundary and fluid variations on the enhancement of diffusive transport. Charles Thompson (Department of Electrical Engineering, University of Lowell, 1 University Avenue, Lowell, MA 01854)

The role that both boundary vibration and unsteady fluid motion have in enhancing diffusive transport will be discussed. In particular, how acoustically induced vortical disturbances interact with a two-dimensional temperature field will be examined. Such vortical disturbances have been shown to be three-dimensional [C. Thompson, "Stability of a Stokes boundary layer," *J. Acoust. Soc. Am.* **81**, 861 (1987)]. Hence, parametric interaction between the flow and temperature field to the major mechanism for diffusion enhancement is expected. Numerical results demonstrating the degree of augmentation will be presented. [Work supported by Analog Devices Professorship and NSF.]

10:35

F7. Noncollinear interaction of a tone with noise. Stephen J. Lind (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029) and Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

The noncollinear interaction of a tone with noise is investigated experimentally in an air-filled rectangular duct. A low-frequency band of noise in the (0,0) mode interacts with a high-frequency pure tone in the (1,0) mode. A quasilinear theory developed by Hamilton and TenCate [*J. Acoust. Soc. Am.* **81**, 1703-1712 (1987)] for the noncollinear interaction of two pure tones is generalized to predict the sum and difference frequency sidebands of noise created around the high-frequency tone. When two pure tones interact, the resulting sum and difference frequency sound oscillates in space with a periodicity that depends on the frequencies and interaction angle of the primary waves. The band of low-frequency noise is thus upshifted to sidebands whose spectral shapes are scalloped in appearance. The scalloping becomes more pronounced with range from the source, and increasing the interaction angle reduces the overall levels of the sidebands. Theory and experiment are shown to be in good general agreement. [Work supported by NSF and by ONR.]

10:50

F8. Propagation and interaction of two collinear finite amplitude sound beams. Jacqueline Naze Tjøtta, Sigve Tjøtta (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway and Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029), and Erlend H. Vefring (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway and Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

The propagation and interaction of collinear, high-intensity sound beams are investigated within the scheme of the parabolic equation

(Khokhlov-Zabolotskaya-Kuznetsov equation). The equation is solved numerically for two monochromatic sources by using a generalization of the method of Hamilton, Naze Tjøtta, and Tjøtta [*J. Acoust. Soc. Am.* **78**, 202-216 (1985)]. Propagation curves and beam patterns for the various components of the acoustic field (harmonics and combination frequencies) are presented. Special attention is given to the difference frequency component. The analysis extends to ranges beyond the shock formation distance. Various source levels, absorption lengths, and downshift ratios are considered. The results are compared with that obtained for sources operating at moderate intensity (quasilinear theory).

11:05

F9. Comparison of tissue composition models via the measurement of the nonlinear parameter of mixtures. Erich Carr Everbach and Robert E. Apfel (Department of Mechanical Engineering, Yale University, 2159 Yale Station, New Haven, CT 06520)

Recently, several models have been proposed for predicting tissue composition (percent water, protein, and fat) from physical measurements made on the bulk material [Sehgal *et al.*, *Ultrasound Med. Biol.* **12**, 865-874 (1986); and R. E. Apfel, *J. Acoust. Soc. Am.* **79**, 148-152 (1986)]. Each model requires the measurement of several bulk properties such as density, sound velocity, and the acoustic nonlinearity parameter B/A . These models are intended to aid the diagnosis and analysis of tissue pathology. They also share with acoustic absorption and nuclear magnetic resonance (NMR) techniques the prospect of elucidating the characteristics of intermolecular interactions in tissues. As reported at a previous meeting [E. C. Everbach and R. E. Apfel, *J. Acoust. Soc. Am. Suppl.* **1** **80**, S4 (1986)], a system to accurately measure the B/A value of liquids and semiliquid substances has been developed. In order to assess the sensitivity of the tissue composition models to differences in molecular structure, B/A has been measured systematically for various proteins and lipids for which acoustic absorption and NMR data are known. To determine the validity of each model, the bulk properties of water-protein-fat mixtures are measured and the model's predictions to the known composition of each mixture are compared. Using the data obtained, the existing models will be compared and suggestions for extending them to more general systems will be offered. [Work supported in part by the U.S. Office of Naval Research and by the National Institutes of Health through Grant 2-R01-GM30419-04.]

11:20

F10. Forced radial oscillations of single cavitation bubbles: A comparison of experimental and numerical studies. G. Holt and L. A. Crum (Physical Acoustics Research Laboratory, Department of Physics, University of Mississippi, Oxford, MS 38677)

Refinements of an optical scattering technique previously reported [*J. Acoust. Soc. Am. Suppl.* **1** **80**, S24 (1986) and *Suppl.* **1** **81**, S26 (1987)] have made possible a quantitative comparison of the experimentally obtained radius versus time ($R-t$) curve with previously obtained numerical $R-t$ results. The response of bubbles with equilibrium radii from 20-80 μ to a 22-kHz acoustic field is presented and compared to the results of previously published numerical models [see, for example, W. Lauterborn, *J. Acoust. Soc. Am.* **59**, 283-293 (1976) and Miksis and Ting, *J. Acoust. Soc. Am.* **81**, 1331-1340 (1987)]. [Work supported by ONR.]

11:35

F11. Focal-point shifts for the reversed wave in acoustical phase conjugation experiments: Paraxial analysis and analogy with optical holography. Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

A phase-conjugate mirror is one that reverses an outgoing wave front

so that it propagates back toward the source. Recent experiments [Kustov *et al.*, *Sov. Phys. Acoust.* **31**, 517-518 (1985)] indicate that wave front reversal can be established through the interaction of a pump wave of frequency f_1 with a signal of frequency f_2 that diverges from a point source. The nonlinear medium where the interaction occurs is a thin layer of gas bubbles in water. In these experiments, however, the frequency f_3 of the reversed wave differed from $f_2 = 60$ kHz since $f_3 = f_1 - f_2 = 40$ kHz and $f_1 = 100$ kHz. The present research is concerned with the use of a paraxial approximation to predict the location of the focal point of the reversed wave. The normals of the wave fronts of the interacting waves are assumed to be nearly perpendicular to the bubble layer. Let z_2 and z_3 denote the respective distances of the source and focal points from the layer. If the pump wave is a plane wave propagating perpendicular to the layer, the analysis gives $z_3 = (f_3/f_2)z_2$, so that $z_3 \neq z_2$ when $f_3 \neq f_2$. This result is analogous to the image location in optical holography for cases where the recording and reconstruction process are carried out at different wavelengths [J. W. Goodman, *Introduction to Fourier Optics* (McGraw-Hill, San Francisco, 1968)]. Other applications of this analogy will be noted. [Work supported by ONR.]

11:50

F12. Blast noise propagation prediction. Carolyn R. Shaffer (Department of Electrical and Computer Engineering, University of Illinois, Urbana-Champaign, Champaign, IL 61820 and USA-CERL, P. O. Box 4005, Champaign, IL 61820-1305), Steve Franke (Department of Electrical and Computer Engineering, University of Illinois, Urbana Champaign, Champaign, IL 61820), and Richard Raspet (Physical Acoustics Research Group, The University of Mississippi, University, MS 38677)

The U.S. Army Construction Engineering Research Laboratory has performed simultaneous meteorological and acoustic measurements of sound propagation from small explosives. Categorized experimental data have been placed into groups based on similarity of variation in sound speed with height. Using these data groups as input to the pulse fast field program [J. Acoust. Soc. Am. **79**, 628-634 (1986)], predictions of sound levels are produced. The comparison of these predictions and measured levels will be discussed.

TUESDAY MORNING, 17 NOVEMBER 1987

TUTTLE NORTH ROOM, 9:00 TO 11:30 A.M.

Session G. Structural Acoustics and Vibration I: Active Vibration Control

Vijay K. Varadan, Chairman

Research Center for the Engineering of Electronic and Acoustic Materials, Pennsylvania State University, University Park, Pennsylvania 16802

Chairman's Introduction—9:00

Invited Papers

9:05

G1. Active control of sound radiation from plates using force inputs. Chris R. Fuller (Department of Mechanical Engineering, Virginia Tech, Blacksburg, VA 24061)

Recently much interest has arisen over the use of active control methods to reduce sound fields. The usual method is to use an array of secondary acoustic sources positioned in the radiated field. This technique has proved somewhat successful, but one difficulty is the excitation of unwanted radiation (control spillover) that necessitates the use of many control sources. A new technique is studied in which the active control is applied directly to the radiating surface by a force input. The advantage of this approach is that, by reducing plate response, the radiated sound levels will be controlled globally. The system considered consists of a baffled clamped circular elastic plate excited on one side by an incident acoustic plane wave. Reduction of the radiated acoustic field on the other side of the plate is achieved by a point force applied directly to the plate surface. The complex amplitude of the control force is determined mathematically such that the radiated acoustic intensity integrated over various required sectors is minimized. Results obtained demonstrate excellent global reduction in the radiated field levels for the single control force and the mechanisms inherent in the method are discussed. [Work supported by NASA Langley.]

9:30

G2. Vibration isolation for spacecraft using the piezoelectric polymer PVF₂. Samuel W. Sirlin (Jet Propulsion Laboratory, Mail Stop 198-326, California Institute of Technology, 4800 Oak Grove Drive, Pasadena, CA 91109)

Throughout the remainder of the current decade and through the 1990s, the pointing requirements of spaceborne scientific payloads will grow increasingly more stringent, yet, for reasons of cost effectiveness, the trend will be away from single-payload-dedicated free-flying spacecraft and toward large, multipayload space vehicles. The basebody structural frequencies of space station/space platforms will be significantly lower than those for previous missions, making the traditional separation between pointing controller bandwidth and structural modes impossible to maintain. Man motion and machine vibration disturbances will likely limit the space station attitude control to relatively coarse levels. In the face of such a demanding dynamic environment,

future space stations will host attached payloads, some of whose pointing requirements approach those of the free-flying space telescope. These issues drive the need for the development of advanced pointing mount technology that can provide a high degree of precision while simultaneously isolating a payload from a disturbance rich host vehicle. A finite element model of an active softmount based on the piezoelectric polymer poly(vinylidene fluoride) has been developed. The model includes the geometric nonlinearities that are associated with the large deflection capability of the softmount. This model is put together with a simple space station model and then both linear frequency domain and time domain simulations are carried out. The wideband disturbance rejection capabilities of the design are demonstrated, both in the frequency domain and in nominal operations.

9:55

G3. The acoustic limit of control of structural dynamics. A. H. von Flotow (Department of Aeronautics and Astronautics, Massachusetts Institute of Technology, Cambridge, MA 02139)

The acoustic limit of active control of structural dynamics is investigated; the limit as the control bandwidth includes a very large number of natural modes of the structure. The point is made that, in this limit, modal analysis cannot provide reasonably accurate models of the structural dynamics and that control design with respect to modal models is then of questionable value. Alternative modeling approaches are reviewed. A particular wave propagation formalism, applicable to modeling the acoustic response of networks of slender structural members, is described in some detail. Control options designed with reference to this formalism are reviewed, and speculations as to future developments of such control are offered.

10:20

G4. Active control of vibrations using externally excited piezoceramic devices. X. Q. Bao, V. K. Varadan, and V. V. Varadan (Research Center for the Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

The design and function of piezoceramic devices containing either one (unimorph) or two (bimorph) piezoceramic disks are described. The function of these devices is to null the transmitted displacement field or both the transmitted and reflected displacement fields when a displacement or pressure field is impressed on one side of the device. Such a device can be used to actively control vibrations of a given frequency by suitably adjusting the voltages that can be applied to one or both the disks. For the unimorph device, fairly wideband conditions of zero transmission can be achieved. For the bimorph device, voltages to the two disks are independently controlled so as to achieve simultaneously the desired conditions of zero reflection and transmission. One or both voltage sources may drain energy from the system. In each case the equivalent electric circuit can be modeled so that the current-voltage characteristics of the device can be calculated and plotted. The piezoceramics are composite in construction containing a piezoelectric phase dispersed in a nonpiezoelectric phase so that the device has the needed mechanical and electromechanical coupling coefficients.

Contributed Papers

10:45

G5. Active vibration control by using the wave control concept. Jiawei Lu and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

A novel active vibration control approach using the concept of wave control is discussed. The idea is to try to change the characteristics of a system and to destroy the condition of resonance response by adding a control force at the boundary of a vibrating system. The approach is different from the modal-space control technique, which is often used in the vibration control of space structures, especially when these structures are built from materials with low damping and are very flexible due to their large size. For the demonstration of this new wave control concept, an active vibration control model for a finite length string is discussed. An exciting force is applied somewhere on the string, and a control force applied at one end of the string is found to absorb the vibrational wave propagating to the end so that no wave reflection appears at the end and the condition of resonance response is destroyed. A beam model has also been developed. The theoretical results are encouraging and better than those obtained by means of the modal-space control approach. The beam

model of active vibration control is being investigated experimentally. [Work supported by SDIO/DNA, Contract No. DNA-001-85-C-0183.]

11:00

G6. Active control of sound fields in elastic cylinders. C. R. Fuller (Department of Mechanical Engineering, Virginia Tech, Blacksburg, VA 24061), R. J. Silcox, and H. C. Lester (Structural Acoustics Branch, NASA Langley Research Center, Hampton, VA 23665)

Much interest has recently arisen over the use of active noise and vibration control in various applications where weight considerations are important. The results of research directed towards applying active methods for reducing noise in the cabins of propeller aircraft will be reviewed. Two differing techniques of implementing the active control will be discussed. The first consists of placing active acoustic sources in the interior space in order to achieve cancellation. The second method relies on active

control of the transmitted acoustical energy by vibrational force inputs applied directly to the cylinder wall. Both theoretical and experimental results using simplified models of aircraft fuselages will be presented and the mechanisms inherent in the control action will be outlined.

11:15

G7. Interaction impedance of a system of pistons coated with an elastic skin using a plane-wave decomposition. Philippe Boissinot (Thomson-Sintra A.S.M., Route des Dolines, Parc d'activités de Valbonne, B.P. 38, 06561 Valbonne Cedex, France)

Expressions of self and mutual impedance of pistons in an infinite, rigid, plane baffle are well known. The array is coated with an elastic rubber skin. Plane-wave decomposition of the compression and shear potentials is performed in the skin as well as the fluid. Expressions are given for the nearfield pressure and for the displacement field in the skin. This pressure is then integrated over the pistons to obtain expressions of self and mutual radiation impedance, which is referred to as average surface velocity. Results are presented for linear (2-D case) and circular (3-D case) radiators in terms of kd (dimensionless product of the wavenumber and the distance between pistons). The other parameters are thickness and elastic characteristics of the skin and the dimension of the piston. As a post processing, the three-dimensional directivity diagram for the whole array can be easily obtained through the plane-wave decomposition in the fluid.

TUESDAY MORNING, 17 NOVEMBER 1987

UM AUDITORIUM LOBBY AND PREFUNCTION AREA, 9:30 A.M. TO 12:00 NOON

Session H. Speech Communication I: Production (Poster Session)

Betty H. Tuller, Chairman

Center for Complex Systems, Florida Atlantic University, Boca Raton, Florida 33431

All posters will be displayed from 9:30 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:30 to 10:45 a.m. and contributors of even-numbered papers will be at their posters from 10:45 a.m. to 12:00 noon.

Contributed Papers

H1. Word-based versus syllable-based models of speech production planning: Evidence from elicited phonological errors. Stefanie Shattuck-Hufnagel (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Earlier error-elicitation experiments have shown that single-segment errors are more likely to be elicited when the target segments share position in the word onset, and less likely when the target segments cross word position (e.g., /p/ and /f/ are more likely to interact in "parade fad" and less likely in "repeat fad," even though both target segments are in pre-stress position in the latter case). A separate series of experiments using CVC stimuli showed a difference in error patterns when the stimulus words are spoken in lists versus embedded in short phrases; stimuli like "leap note nap lute" elicit about the same proportion of errors in initial and final position, while the phrasal version ("From the leap of the note to the nap of the lute") elicits far fewer errors in final position. If this difference between word lists and phrases extends to other aspects of error patterns, it might affect the word-position/stress-position results for mixed one- and two-syllable stimuli. However, results show that, when these stimuli are embedded in phrases, the error patterns remain remarkably similar. Moreover, when speakers are asked to generate sentences using the words of the stimulus, instead of reading or reciting the phrases provided by the experimenter, results again support the claim that word-onset consonants are more likely to interact in an error than are pre-stress consonants. These findings suggest that word structure plays a role in the production processing representation that is in force when single-segment errors occur.

H2. Task-dynamic modeling of interarticulator coordination. Elliot L. Saltzman, Philip E. Rubin, Louis Goldstein, and Catherine P. Browman (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Speech production involves the formation and release of constrictions at different points in the vocal tract. Progress is reported on a *task-dynamic* computational model of autonomous coordination among the components of articulatory synergies (e.g., lips and jaw) that participate in motions along functionally defined *tract variables* (e.g., lip aperture and protrusion). For each speech gesture, a time-invariant dynamical system (damped, second-order) is specified at the tract-variable level, and is transformed into a gesturally and posturally specific dynamical system for the synergy components. Articulatory movement patterns emerge as implicit consequences of the gesture-specific tract-variable control structures and the ongoing postural state of the articulators. Explicit trajectory planning is not required. Significantly, the modeled coordinative processes are exactly the same during simulations of unperturbed, mechanically perturbed, and coproduced speech gestures. All simulations are implemented using the Haskins Laboratories software articulatory synthesizer, and are restricted to the vocal tract's sagittal plane and the simplified articulatory geometry represented in the synthesizer. [Work supported by NIH Grant NS-13617 and NSF Grant BNS-85-20709.]

H3. Source models for fricative consonants. Christine H. Shadle (Department of Electronics and Computer Science, University of Southampton, Southampton SO9 5NH, England)

Source parameters for models of fricative consonants were derived from experiments using mechanical models having the profile of midsagittal x rays given in Fant [*Acoustic Theory of Speech Production* (Mouton, The Hague, 1970)]. By positioning a pressure probe at several points along the wall of each model's tract, it was shown that sound-generation characteristics of fricatives such as /s, ʃ/ differ fundamentally from those of /ç, x/ with regard to source location, source spectrum level at low frequencies, and the relationship between source spectrum amplitude and mean volume velocity. In both cases, sound is generated when turbulent

air strikes relatively rigid boundaries. The angle of the boundary with respect to the flow of air is the significant factor in this case. For /s, f/, the obstacle is at right angles, for /ç, x/, the "obstacle," the tract wall, is nearly parallel to the flow. Implications for vocal tract models will be discussed. [Work supported by a Hunt Fellowship awarded by the Acoustical Society, and a grant awarded by the North Atlantic Treaty Organization in 1984.]

H4. A twisting one-mass glottal model. Lloyd Rice (Phonetics Laboratory, UCLA, Los Angeles, CA 90024-1543)

A new model of vocal cord vibration will be presented that represents the cord body by a single mass exhibiting oscillatory behavior on a horizontal axis together with a rotational component. The model is adequate to describe the diverging and converging configurations of the glottal opening as observed at different phases of the glottal cycle. Characteristics of the model are explored under various glottal vibratory modes and are compared with those of existing models such as the popular two-mass configuration.

H5. Bending-beam model of vocal-fold vibration. Corine Bickley (Department of Electrical Engineering and Computer Science and Research Laboratory of Electronics, Room 36-521, Massachusetts Institute of Technology, Cambridge, MA 02139) and Kevin Brown (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Recent theoretical developments have shown that a model based on the theory of bending beams is the best predictor of the fundamental frequencies of young children's speech [Bickley, Proc. 11th Int. Congr. Phon. Sci. (1987)]. For children of age 3 years and younger, the values predicted by a spring-mass model are too low, and the values of a vibrating-string model are too high. In the current study, the effect of growth of the vocal-fold structure on fundamental frequency is analyzed. The vocal-fold structure grows nonuniformly; for example, a 1-year-old's vocal folds are approximately one-fifth as long as an adult's folds but only 10% thinner. The bending-beam model is an improvement over other models because it reflects the structure of the vocal folds and the attachments of the vocal-fold tissue to the arytenoid and thyroid cartilages, characteristics previously ignored. The fundamental frequency of the new model depends on both the beamlike characteristics of the vocal folds and the springiness of the structure. Changes in fundamental frequency are predicted from the model as a function of age.

H6. Estimating vocal tract shapes from x-ray microbeam data. Peter Ladefoged and Mona Lindau-Webb (Phonetics Laboratory, UCLA, Los Angeles, CA 90024-1543)

X-ray microbeam techniques provide data on a few points on the tongue. The possibility of an algorithm that will turn these data into a full specification of the whole tongue shape is tested. A weakness of the system is that it is difficult to place pellets on the posterior half of the tongue. The first goal is thus to investigate what is the minimum number of pellets necessary to predict the shape of the anterior part of the tongue. Data were recorded from seven points, four of which were on the front of the tongue, approximately 15 mm apart. The subject produced short utterances illustrating English vowels. This procedure was then repeated with the four tongue pellets moved approximately 5 mm forward. Image scans of sustained English vowels in which the subject had a chain on the tongue midline consisting of 17 pellets 5 mm apart were also recorded. The results allowed determination of the extent to which tongue shapes can be determined from x-ray microbeam pellets. [Work supported by NIH.]

H7. Voicing distinction for fricatives: Acoustic theory and measurements. Kenneth N. Stevens (Department of Electrical Engineering and Computer Science and the Research Laboratory of Electronics, Speech Communication, Massachusetts Institute of Technology, Cambridge, MA 02139), Sheila E. Blumstein (Department of Cognitive and Linguistic Sciences, Brown University, Providence, RI 02912), and Laura B. Glicksman (Research Laboratory of Electronics, Speech Communication, Massachusetts Institute of Technology, Cambridge, MA 02139)

Theoretical analysis suggests that simultaneous generation of vocal-fold vibration and turbulence noise at a supraglottal constriction requires rather precise adjustment of the structures controlling the glottis, the intraoral volume, and the supraglottal constriction. The theory shows that there tends to be a reciprocal relation between voicing amplitude and the amplitude of the turbulence noise. It is not surprising, therefore, that vocal-fold vibration does not always occur throughout the constricted interval when a "voiced" fricative is produced, and that variation in the noise amplitude is often observed. Measurements of the amplitude of voicing (expressed as the spectrum amplitude in the region of the first harmonic) and of friction noise have been made for several fricative consonants spoken by several speakers in a number of phonetic environments, including utterance final and intervocalic position, and in clusters with other voiced and voiceless fricatives. Evidence for voicing can usually be found over some portion of the constricted interval, particularly in the vicinity of fricative onset or release. However, voicing may be weak or absent when the voiced fricative is followed in a cluster by a voiceless fricative. [Work supported in part by Grants NS-04332 and NS-15123 from the National Institutes of Health.]

H8. Stress effects on stop consonant closure and release gestures. Mark K. Tiede (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

This study seeks to characterize the articulatory kinematics of stop consonant gestures in contrasting environments, with the aim of incorporating the results into the gesture-based approach to synthesis under development at Haskins. The CVCVC tokens drawn from the Bell x-ray microbeam corpus have been examined, representing combinations of stress, vowel, and consonant type spoken in isolation by a female speaker of English. Measurements were taken from peaks of the vertical component of pellet displacement and velocity corresponding to consonant closure and release gestures, using the lower lip pellet for labials and the tongue root pellet for velars. The ANOVA results show that stress affects consonant gesture duration independently of coproduced vowel type, but interacts significantly with consonant type, possibly because velars compete with coproduced vowels for control of the tongue. Results also showed different results for consonant duration depending on whether the gesture occurred in the first or final syllable, which is consistent with previous studies of prepausal lengthening [D. H. Klatt, J. Phonet. 3, 129-140 (1975)]. [Work supported by NSF Grant BNS-8520709.]

H9. Interaction of speaking rate and postvocalic consonantal voicing on vowel duration in American English. H. S. Gopal (Callier Center, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235) and Ann K. Syrdal (AT&T Bell Labs, IH6C-320, Naperville-Wheaton Road, Naperville, IL 60566)

Vowel duration is influenced both by speaking rate and by voicing of the postvocalic consonant. Our study systematically investigates how these two factors interact. Syllable and segment durations were measured for /pVC/ target syllables containing one of eight vowels and voiced or voiceless stops or fricatives. Target syllables were produced in sentence contexts at three speaking rates by four female and three male native American English speakers. Our findings replicate and extend Klatt's incompressibility model of vowel duration [D. H. Klatt, J. Acoust. Soc. Am. 54, 1102-1104 (1973)]. The two factors influencing vowel duration do not combine independently in a multiplicative or additive fashion. Instead, they interact such that vowels manifest a resistance to further shortening.

H10. Token-to-token variation of tongue-body vowel targets: The effect of coarticulation. J. S. Perkell and M. H. Cohen (Research Laboratory of Electronics, Room 36-543, Massachusetts Institute of Technology, Cambridge, MA 02139)

Variation of vowel targets for a midsagittal point on the tongue body of a single speaker of American English was examined. The subject pronounced multiple repetitions of nonsense utterances of the form /bV(C)VbA/, in which the vowels were /i/, /u/, /a/ and the consonant, when present, was /b/ or /h/. Stress was placed on the second syllable. Movements of a single point on the middle of the tongue-body dorsum were transduced with a new alternating magnetic field movement transducer system. For each vowel, a scatter plot was generated of "target" positions (x, y coordinates) at the moment of minimum tangential velocity. In general, these scatter plots are elongated. When the vowels in the first and second syllables are the same, the scatter in target position is relatively tighter and less elongated than when the vowels are different. When the vowels in the two syllables are different, the long axis passing through the scatter for one vowel is oriented in the direction of the target for the other vowel, providing a "statistically based" demonstration of context dependence of articulatory targets. [Work supported by NIH Grant No. NS04332.]

H11. Fundamental frequency and intensity control in speech. Helmer Strik and Louis Boves (Institute of Phonetics, Nijmegen University, P. O. Box 9103, 6500 HD Nijmegen, The Netherlands)

A gradual downdrift of fundamental frequency (F_0) and intensity (IL) during speech utterances has been observed in almost all languages. Because both F_0 and IL are related to subglottal pressure (P_{sg}) and because P_{sg} is supposed to decrease as the air supply in the lungs decreases, it has been hypothesized that this downdrift, usually called declination, is a passive process. In utterances where declination is absent, like in some types of questions, active countermeasures would then be necessary. In this research, P_{sg} , airflow, electroglottogram, speech signal, and EMG activity of the cricothyroid, sternohyoid, and vocalis muscles have been recorded, while linguistically naive subjects sustained vowels on a fixed F_0 and IL and produced a number of sentences with different intonation patterns. Results indicate that there is no direct relation between air supply in the lungs and P_{sg} , and that P_{sg} , F_0 , and IL can be kept on a desired level without active involvement of the muscle systems that control intonation in speech. [Research supported by the Foundation of Linguistics, funded by Z. W. O.]

H12. Respiratory response to loss of velar resistance. Kathleen E. Morr, Anne H. B. Putnam, and Donald W. Warren (Dental Research Center, University of North Carolina, Chapel Hill, NC 27514)

The hypothesis that speech aerodynamics conforms to the principles of a regulating system was tested in nine normal adult subjects. The velar mechanism was perturbed by having subjects lower the soft palate during a series of words involving plosive consonants. The pressure-flow technique was used to measure oral pressures and velar resistance. Inductive plethysmography was used to measure tidal volumes associated with test consonants. The data indicate that intraoral pressure remained at appropriate levels (> 4.0 cm H₂O) after loss of velar resistance. Tidal volume did not change during inspiration but increased during speech expiration. The difference was statistically significant ($p < 0.01$). These results support previous findings of Warren [Cleft Palate J. 23, 251-260 (1986)] in cleft subjects, Putnam *et al.* [J. Speech Hear. Res. 29, 37-49 (1986)] in perturbation studies, and Warren *et al.* [J. Acoust. Soc. Am. 67, 1828-1831 (1980), Folia Phoniatr. 33, 380-393 (1981), Folia Phoniatr. 36, 164-173 (1984)] in bite block studies that intraoral pressures are maintained at appropriate levels in the presence of lowered airway resistance. It is concluded that, under the test conditions, pressure was regulated by controlling respiratory effort. This compensatory response increased airflow rate and volume when resistance fell. [Work supported by NIDR.]

H13. Stress effects on velic gestures. Rena Arens Krakow (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

In his cross-language survey of oral and nasal vowel patterns, Schourup [Work. Pap. Linguist. 15, 190-221 (1973)] reported that contextual vowel nasalization is more likely to occur in stressed syllables than in unstressed syllables. This study looks at how spatial and temporal patterns of velic movement are affected by stress and finds articulatory evidence in support of Schourup's claim. Vertical movements of the velum were monitored while the (American English) speaker produced utterances with different stress patterns (mVbVb vs mVbVb; bVbVm vs bVbVm). Velic height measured at the midpoint of the vowels adjacent to the nasal consonants was significantly lower when the vowels were stressed. This pattern, however, primarily reflected temporal differences in the velic gestures due to stress rather than changes in the spatial extent of velic lowering for the nasal consonants. As an example, the velic lowering gestures for the initial /m/ in /mábab/ and /mábáb/ appeared quite similar both in their timing and spatial extent, but velic raising from the low position was initiated considerably later in /mábab/ than /mábáb/. These findings indicate that part of the articulatory distinction between syllables with different levels of stress is the temporal pattern of velic movement. [Work supported by NIH Grants NS-13617 and HD-1994.]

H14. Generating intonational support for discourse. James Raymond Davis (AT&T Bell Laboratories, Murray Hill, NJ 07974 and Massachusetts Institute of Technology, Media Technology Laboratory, Cambridge, MA 02139)

Proper intonation can make a spoken message more intelligible by marking information as old or new and communicating the hierarchical structure of information. Use of intonational variation is especially feasible in a system that synthesizes speech from an abstract representation (instead of from text) because the program already knows the structure of the information to be conveyed. This paper presents a schema for assigning pitch range and choosing pitch accents. The schema draws upon work on discourse [B. J. Grosz and C. L. Sidner, Comput. Linguist. 12, 175-204 (1986)] and on intonational meaning [J. Hirschberg and J. Pierrehumbert, Proc. 24th Meet. Assoc. Comput. Ling., 136-144 (1986)]. The model has been tested with a program which generates spoken directions for driving in Cambridge [J. Davis, Proc. Am. Voice I/O Soc. (1986)].

H15. Evaluation of speech under stress and emotional conditions. John H. L. Hansen and Mark A. Clements (School of Electrical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This paper presents results from an investigation of how stress and emotion affect speech characteristics with specific application to improving automatic speech recognition. Past studies have been limited in scope, often using only one or two subjects and analyzing only one or two parameters (typically involving pitch). A comprehensive speech under stress database has been established at Georgia Tech for the purposes of stress research. The database is partitioned into five domains, encompassing a wide variety of stresses that include: various talking styles (slow, fast, soft, loud, angry, clear, question, in noise), single and dual tracking workload stress inducing tasks, emotional speech from psychiatric analysis session, and subject motion-fear tasks. A total of 32 speakers was employed to generate in excess of 16 000 utterances. The database evaluation was partitioned into three areas. Analysis was first performed on (i) *speech with simulated stress* and (ii) *speech from stress inducing workload tasks or speech in noise*. Statistically significant parameters were established, and an equivalent analysis was carried out over (iii) *speech produced under actual stress and emotion*. This scheme was chosen since simulated conditions allowed for careful control of vocabulary, task requirements, and background noise characteristics. Evaluation over actual stress or emotional conditions was used to verify results established under simulated conditions. Variables employed in the evaluation include: pitch (mean, variance, higher moments, point process characterization), glottal waveform characteristics, glottal source spectrum (spectral tilt, energy concentration), duration, intensity, formant locations and bandwidths, vo-

cal-tract shaping based on line spectral pair coefficients, and the variability of each of these. Resulting categorizations will be presented, as well as the statistical significance of each set of measurements. [Research supported by U. S. Army Human Engineering Labs.]

H16. The control of voice onset time and vowel duration after paraneoplastic cerebellar degeneration: An acoustic study. Susan J. Behrens, Neil Anderson, Jerome Posner, and John J. Sidtis (Department of Neurology, Memorial Sloan-Kettering Cancer Center, 1275 York Avenue, New York, NY 10021)

Speech production was examined in a group of six subjects with paraneoplastic cerebellar degeneration, a neurological disorder that produces severe dysarthria. Subjects were recorded repeating CVC and CVCV syllables that included the stop consonant [p b t d k g] and the vowels [i a u]. Vowel durations preceding voiced and voiceless segments and voice onset time (VOT) were measured. The vowel duration measures assessed the preservation of intrinsic duration properties, and the variation of these properties in conditioning phonological contexts. The VOT measure assessed the coordination of two independent articulatory events. Results indicated that the cerebellar subjects normally differentiated among individual vowels, and altered vowel duration as a function of phonological context. In contrast, they tended to reduce VOT for voiceless segments, and, in the case of the most impaired subject, the voiceless segments were always perceived as being voiced. Although the VOT durations were reduced, there was no overlap between voiced and voiceless VOTs for most cerebellar subjects. These results suggest that the temporal coordination of articulatory maneuvers rather than general timing of segmental duration may specifically involve the cerebellum. [Work supported by NS17778.]

H17. The control of voice onset time and vowel duration after paraneoplastic cerebellar degeneration: Correlation with regional cerebral glucose metabolism using positron emission tomography. John J. Sidtis, Susan J. Behrens, James R. Moeller, Stephen C. Strother, Neil Anderson, Jerome Posner, and David A. Rottenberg (Department of Neurology, Memorial Sloan-Kettering Cancer Center, New York, NY 10021)

In a companion report, an impairment in voice onset time (VOT) control was described for a group of subjects with paraneoplastic cerebellar degeneration who could otherwise produce appropriate segmental duration. The underlying neurophysiological changes were related to regional cerebral metabolic rates for glucose (rCMRGlu) determined by F18-fluorodeoxyglucose/positron emission tomography (PET). Nine cerebellar subjects and 18 normal controls were studied with eyes patched while listening to music and periodic verbal briefings. The rCMRGlu was calculated for each of 28 anatomically defined regions of interest (ROI). The cerebellar subjects could be differentiated from normals by their lower global metabolic rate, and by their rCMRGlu pattern. For those cerebellar subjects who could participate in the acoustic study, cerebellar rCMRGlu was significantly below normal range ($p < 0.05$) in all but one case. Severity of VOT abnormality appeared to be related not only to cerebellar rCMRGlu, but to abnormally low cortical rCMRGlu as well. The role of the cerebellum in maintaining temporal coordination of articulatory maneuvers appears to involve the coordination of cortical and subcortical brain regions. [Work supported by NS17778 and NS23473.]

H18. Acoustic analysis of hoarse voices in running speech. Hiroshi Muta, Thomas Baer (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), and Hiroyuki Fukuda (Department of Otolaryngology, Keio University School of Medicine, 35 Shinanomachi, Tokyo 160, Japan)

A method of pitch-synchronous acoustic analysis of hoarseness requiring a voice sample of only four fundamental periods is presented. This

method calculates a noise-to-signal (N/S) ratio, defined from the power spectrum, which indicates the depth of valleys between the harmonic peaks. A discrete Fourier transform based on a continuously variable window spanning exactly four fundamental periods is used to calculate the spectrum. A two-stage procedure is used to determine the exact duration of the four fundamental periods. An initial estimate is obtained using autocorrelation in the time domain. A more precise estimate is obtained in the frequency domain by minimizing the errors between the DFT and the predicted spectrum of a windowed harmonic signal. Analysis of synthesized voices shows that the N/S ratio is sensitive to additive noise, jitter, and shimmer, and is insensitive to slower (8 Hz) modulation in fundamental frequency and amplitude. An analysis of pre- and post-operative running speech of six patients with laryngeal polyps or nodules has consistently indicated the successful therapeutic results for all subjects. [Work supported by NIH Grant NS-13870.]

H19. Fundamental frequency contours produced by normal and dysarthric speakers. Helen Southwood and Gary Weismer (Department of Communicative Disorders, University of Wisconsin—Madison, Madison, WI 53706)

Perceptual descriptions of dysarthric speech often include terms that are meant to index aberrancy in sentence-level fundamental frequency (F_0) contours. Whereas there is general agreement that many dysarthric speakers produce "abnormal" F_0 contours, two relevant issues that appear to be unanswered are: (1) How should "normal" F_0 contours be characterized, and (2) what is the nature of F_0 contours produced by dysarthric speakers. In the present paper, a report is made on a category approach to identifying ranges of normal F_0 production in young adults and geriatrics, and then F_0 contours of speakers with hypokinetic (due to Parkinson's disease) and spastic (congenital) dysarthria are shown. Discussion will focus on the success of the category system, and the motor control implications of the dysarthric F_0 contours. [Work supported by NIH.]

H20. Acoustic phonetic features of young children's potential homophones. George D. Allen, Diane Frome Loeb, Lori Swanson, Laurence B. Leonard, and Richard G. Schwartz (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

Homophones are different words that sound alike, e.g., "bear" and "bare." Since young children can often hear differences they cannot speak, they may produce homophones for words they know to be different. Thus, for example, a child may hear that "bike" and "bite" are different, but if s/he cannot yet produce word-final consonants, s/he may utter both as /bai/. Two questions presently under investigation are: (1) Do young children actively seek to include or exclude such potential homophones in their early vocabulary? (2) Do children faced with such potential homophones produce subtle, subphonemic differences between them? This paper reports the results of a study aimed at answering the second of these two questions. Nine 20-month-old normally developing, English-speaking children's utterances were tape recorded for acoustic analysis. These utterances were produced in situations in which homophones were likely to result from (a) reduction of initial consonant clusters, (b) stopping of initial fricatives, or (c) deletion of final consonants. Although it is hypothesized that the most likely sources of subphonemic differentiation would be VOT [for type (a) and (b) pairs] and vowel duration [for type (c)], F_0 and formant frequencies were also measured and detailed auditory phonetic analyses were performed. Results from these analyses will be discussed in terms of individual and group patterns of differentiation between potentially homophonous word pairs. [Work supported, in part, by NIH.]

H21. Cycle to cycle spectral perturbations in the voices of female speakers. A. Yonovitz and L. Yonovitz (Speech, Language and Learning Center, 12841 Jones Road, Houston, TX 77070)

The glottal function was obtained through the use of a reflectionless tube. This tube acted as a pseudoinfinite termination of the vocal tract. Ten female speakers phonated a neutral vowel while target matching to a 210-Hz tone. The glottal waveform was digitized and each cycle was partitioned at zero crossings. This procedure provided an accurate determination of the period and separate timing measures for the positive and negative portions of the glottal pulse. Discrete Fourier analysis on each individual cycle indicated changes in the spectral content. The data showed a systematic change in the spectral parameters related to the open-close ratio for each cycle. Spectral perturbation analysis may be a more inclusive measure of waveshape changes than jitter or shimmer.

H22. The timing of prenuclear high accents in English. Kim E. A. Silverman (AT&T Bell Laboratories, Room 2C-440, 600 Mountain Avenue, Murray Hill, NJ 07974) and Janet B. Pierrehumbert (AT&T

Bell Laboratories, Room 2D-452, 600 Mountain Avenue, Murray Hill, NJ 07974)

In English, the alignment of intonation peaks with their syllables exhibits a great deal of contextually governed variation. Understanding this variation is of theoretical interest, and modeling it correctly is important for good quality intonation synthesis. An experimental study of the alignment of prenuclear accent peaks with their associated syllables will be described. Two speakers produced repetitions of names of the form "Ma Lemm," "Mom LeMann," "Mamalie Lemonick," and "Mama Lemonick," with all combinations of the four first names and three surnames. Segmental durations and the F_0 peak location in the first name were measured. Results show that although both speaking rate and prosodic context affect syllable duration, they exert different influences on peak alignment. Specifically, when a syllable is lengthened by a word boundary (e.g., Ma Le Man versus Mama Lemm) or stress clash (e.g., Ma Lemm), the peak falls disproportionately earlier in the vowel. This seems to be related to the syllable-internal durational patterns. It is concluded that rules for generating phonetic details from phonological structure must access information about the upcoming prosodic context.

Session I. Physical Acoustics II: Ultrasonics and Physical Effects of Sound

Yves H. Berthelot, Chairman

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

Chairman's Introduction—1:00

Contributed Papers

1:05

I1. Some experiments in ultrasonic pulse shaping. D. Kent Lewis (Lawrence Livermore National Laboratory, NDE Section, Livermore, CA 94950) and Bill D. Cook (University of Houston, Cullen College of Engineering, Houston, TX 77004)

Consider a pulse/echo ultrasonic system where the ultrasound is reflected from a large parallel reflector. Control of the shape of the output electrical signal out of this system is attempted by shaping the electronic pulse initially applied to a transducer. Analysis shows that transduction, both generation and reception, and propagation each involve differentiation. Instead of having the goal to be a unipolar pulse, the desired waveform has been chosen to be a third derivative of a Gaussian function. Initially, a Gaussian-shaped pulse is sent into the system, and it is determined by signal processing, what modifications of the driving waveform are to be made, and then this newly shaped pulse is sent to the transducer. The output is then very close to our desired third derivative of a Gaussian. Integration of the output signal three times yields a unipolar pulse of Gaussian shape with very small kurtosis.

1:20

I2. Application of acousto-optic light modulation to the laser generation of ultrasound. Jacek Jarzynski and Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

When a laser beam passes through an acousto-optic Bragg modulator operated at constant frequency, one observes the splitting of the laser beam into several directions at angles proportional to the input frequency of the Bragg cell. Consequently, scanning of a laser beam can be achieved purely electronically by applying an FM chirped signal to a Bragg modulator. A scanning rate of about $4 \text{ mm}/\mu\text{s}$ has been observed at a distance of 3 m from a Bragg cell modulated between 35 and 40 MHz in a 4- μs interval with a 1-W cw argon-ion laser source. Such a scanning technique can be used to enhance the thermal generation of ultrasonic waves in a sample illuminated with a laser beam by scanning the laser beam at a velocity that matches precisely the wave velocity in the sample. Another improvement under investigation is to use multimode optical fibers in order not only to increase the range of scanning velocities, but also to control precisely the light distribution on the surface of the sample. This noncontact method of laser generation of ultrasonic waves may find some applications in nondestructive testing.

1:35

I3. Ultrasonic wave propagation in multilayered fibrous composites. Adnan H. Nayfeh (Department of Aerospace Engineering, M.L. 70, University of Cincinnati, Cincinnati, OH 45221) and Dale E. Chimenti (Materials Laboratory, Wright-Patterson Air Force Base, Dayton, OH 45433)

A unified analytical treatment supported by extensive experimental data of the interaction of ultrasonic waves with single and multilayered

fibrous composite plates is presented. The plates are supposed to be immersed in water and subjected to incident waves at arbitrary angles from the normals. For the single laminate plate, solutions are presented for arbitrary azimuthal angles. However, for the general multilayered plate case, treatment is restricted to the case where the wave is incident along planes of symmetry of each individual lamina. Reflection and transmission coefficients are derived, from which characteristic behavior is identified. Comparisons between theory and experiments in the form of reflected energy distribution and identification of dispersion based upon a total transmission criterion are given.

1:50

I4. Measurements of ultrasonic parameters in fluid-filled porous metals with different grain size. Laszlo Adler, Qiang Xue (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210), Alain Jungman, and Gerard Quentin (Groupe de Physique des Solides, Universite Paris 7, Paris, France)

Ultrasonic wave propagation in sintered metallic samples (immersed in fluid) was investigated. Samples made out of bronze and out of stainless steel materials with particle sizes ranging from 30–600 μ were used in this study. In order to correlate particle size with ultrasonic parameters, all samples were chosen with approximately the same volume porosity of 30%. Fast and slow compressional waves, surface wave velocities, as well as attenuation, were measured. The frequency dependence behavior of these ultrasonic parameters was studied from 1–20 MHz and correlated to particle size. [This work was supported by NATO Grant 0131/87 and by the National Science Foundation Grant INT-8614592.]

2:05

I5. Quantitative particle characterization by ultrasound. Ronald A. Roy and Robert E. Apfel (Yale University, P.O. Box 2159, New Haven, CT 06520)

An acoustic scattering technique was developed for determining the compressibility and density of microparticles (diameter less than 15 μm), which were modeled as fluid spheres. Tone bursts of 2- μs duration and 30-MHz center frequency were scattered from individual microparticles as they traversed the confocal region of two transducers. The scattered bursts were detected at 90° and 180° (backscattering), digitized, and stored in computer memory. By using either the Rayleigh or Born approximations, it was possible to determine the compressibility and density of the particles given *a priori* knowledge of the particle size and host properties. It was subsequently possible to compute the mean compressibility and density averaged over a population of particles, or to display histograms of scattered amplitude statistics. An experiment was run to assess the feasibility of using polystyrene microspheres to calibrate the instrument. A second study was performed on the buffy coat harvested from whole human blood. Finally, Chinese hamster ovary cells were studied in order to see if the instrument could detect hyperthermia induced membrane blistering. [Work supported by NIH and ONP.]

2:20

16. Dependence on pulse parameters for free radical production by cavitation from short pulsed ultrasound. J. B. Fowlkes and L. A. Crum (National Center for Physical Acoustics, P.O. Box 847, Fraternity Row, University, MS 38677)

Recent data will be presented that indicate the presence of free radicals generated with peak acoustic pressure amplitudes and pulse shapes characteristic of some medical diagnostic devices. The free radicals were detected via a chemiluminescence reaction in an aqueous solution containing luminol using a single photon counting technique. The acoustic pressure threshold for light emission was examined as a function of pulse width, duty cycle, and pulse repetition frequency (PRF); possible explanations for these dependences based upon a simple concept of cavitation inception will be presented. [Work supported by the National Institutes of Health.]

2:35

17. Interaction of Lamb waves with surface-breaking cracks in centrifugally cast stainless steel (CCSS) plates. Tran D. K. Ngoc and Kin W. Ng (Department of Physics, Georgetown University, Washington, DC 20057)

Application of conventional ultrasonic nondestructive evaluation techniques to CCSS components in nuclear power plants has been limited principally due to the anisotropy of CCSS materials. Phenomena such as beam skewing and distortion are directly attributable to this anisotropy and cause severe difficulties in defect detection and sizing. The present investigation is intended to use Lamb waves as the probing mechanism to detect and characterize a surface-breaking crack residing on an interior surface not accessible by the ultrasonic probe. Experimental investigation is directed at several techniques for exciting and detecting Lamb waves in this application. The excitability of various Lamb-wave modes in the fd (frequency times thickness) range of interest and the effects of anisotropy and surface curvature on mode propagation characteristics are also studied. Signals due to mode conversion and crack diffraction are used to develop an interpretation procedure for crack detection and sizing. [Work supported by EPRI under contract RP2405-23.]

2:50

18. On a plate/surface wave mode selection criteria for ultrasonic evaluation in layered structures. A. Pilarski, J. L. Rose, and K. Balasubramaniam (Department of Mechanical Engineering and Mechanics, Drexel University, Philadelphia, PA 19104)

The choice of specific modes while utilizing plate or surface waves, in the ultrasonic evaluation of layered structures, is extremely critical, especially if the purpose is to tune our sensitivity to particular types of imperfections. A direct approach based on the analysis of each individual mode, for particular geometrical configuration and frequency content, is very time consuming. Therefore, it has been proposed [A. Pilarski, *Arch. Acoust.* 7, 61-70 (1982)] to use an indirect approach by comparing the dispersion curves for different material properties, geometrical constraints, boundary conditions, etc. In this paper, this technique has been applied to a three-layered aluminum-epoxy-aluminum adhesively bond-

ed structure. This was achieved through the numerical solution of the characteristic equation for such a layered structure. The results thus obtained were verified through experimentally generating and studying the behavior of plate/surface waves in specially prepared Al-epoxy-Al bonded specimens with adhesive weaknesses. Both immersion and contact methods are applied in our experimental investigations.

3:05

19. Acoustically generated temperature gradients in plates. Michael Muzzerall, Anthony A. Atchley, and Thomas J. Hofer (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The results of measurements of thermoacoustically generated temperature gradients in short thin plates located in a resonant tube are reported. The temperature gradient, resulting from a heat flow generated by the acoustic field, is a function of the acoustic pressure amplitude, the Prandtl number of the gas, the configuration of the plates, and the position of the plates in the tube. Measurements were made for pressure amplitudes ranging from approximately 145-162 dB re : 20 μ Pa, in argon and helium, for single plates and stacks of up to five plates having separations from approximately 4 to 40 thermal penetration depths, and at various positions in the tube. The results are compared to predictions based on a theory by Wheatley and others [J. Wheatley *et al.*, *J. Acoust. Soc. Am.* 74, 153-170 (1983)]. For pressure amplitudes below 150 dB, the measurements agree well with theory. At higher pressure amplitudes, the agreement diminishes, indicating the presence of effects not addressed by theory. [Work supported by NPS Foundation.]

3:20

110. Possibility of improved laser-Doppler detection of sound by tracking moving phase fronts. Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Laser-Doppler anemometry applied to acoustics presents special problems because of the small velocities of fluid flow associated with sound and because of the relatively rapid oscillations. The analysis of Hanish, attributed to General Electric Company (1967) and demonstrated by K. J. Taylor in 1976 and 1981, shows such problems can be overcome in major part for single frequency sound using the spectral density of the photomultiplier tube output. Comparable success is unlikely for broadband sound; an "uncertainty principle" requires averaging over time intervals comparable to the reciprocal of the Doppler shift in frequency; no black box electronics can beat this principle. Difficulties appear insurmountable if characteristic acoustic frequencies are much higher than the Doppler shift. A possible way of overcoming this is to take advantage of the constancy of the sound speed and of the omnidirectionality of incoming sound. Optical systems can be designed which move the effective optical scattering volume along nearly rectilinear paths at the sound speed such that the laser tracks a moving phase front. The fluid velocity sensed by the laser thus appears to be constant rather than oscillating, so one has the opportunity to examine meaningful averages over sufficiently long time periods. [Work supported by ONR.]

Session J. Speech Communication II and Bioresponse to Vibration I: Special Focus on Speech Perception Using Tactile Aids

Rebecca Eilers, Chairman

Mailman Center for Child Development, University of Miami, P. O. Box 016820, Miami, Florida 33101

Chairman's Introduction—1:00

Invited Papers

1:05

J1. Evaluations of single-channel and multichannel tactile aids for the hearing impaired. Janet M. Weisenberger (Central Institute for the Deaf, 818 S. Euclid, Saint Louis, MO 63110)

A number of single-channel and multichannel tactile devices for hearing-impaired persons have been evaluated in our laboratory over the last several years. Both commercially available and experimental devices have been tested, including the Tactaid I, II, and V, Minifonator, Minivib, Queen's University tactile vocoder (both laboratory and wearable versions), and Tacticon TC-1600. Normal-hearing and hearing-impaired adults, and hearing-impaired children, have participated in a variety of training tasks, employing both recorded and live-voice stimuli. These tasks include simple detection of sound, environmental sound identification, syllable rhythm and stress categorization, phoneme identification, word identification, phrase and sentence identification, connected discourse tracking, and a question-and-answer "conversation" task. In the present paper, the results from these evaluations are considered as a whole, to permit generalizations about the kinds of information that can be provided by single-channel, dual-channel, and multichannel devices, with and without the addition of lipreading. In addition, results are discussed in terms of the development of "optimal" training procedures for the use of tactile aids. [Work supported by NSF and NIH.]

1:35

J2. Speech perception through the sense of touch by profoundly deaf adults and children. Michael P. Lynch, Rebecca E. Eilers, and D. K. Oller (Departments of Psychology and Pediatrics, University of Miami, Mailman Center for Child Development, P. O. Box 016820, Miami, FL 33101)

Eight congenitally, profoundly deaf children and two deaf adults received unimodal vocabulary recognition training with either a multichannel electro-tactile aid or a two-channel vibrotactile aid. The children learned to recognize vocabularies of 15–20 words with 70%–80% accuracy. The adults learned a 50-word list. Following this training, subjects' identification of familiar and novel words was assessed using the tactile devices and aided audition. Vocabulary recognition was tested in each of three conditions: (1) with aided audition, (2) with dual or multichannel aid, and (3) with both tactual and auditory aids. Results indicate that auditory and tactual cues were successfully integrated by both children and adults to yield significantly better performance in combined than in single modality conditions. An additional study of the narrative tracking performance of one of the deaf adults revealed a similar synergistic interplay of taction, audition, and lipreading. [Work supported by Rita and Jerome Cohen, Renata Mahan, and Austin and Marta Weeks.]

2:05

J3. Lipreading with single-channel vibrotactile presentation of voice fundamental frequency. Lynne E. Bernstein, Silvio P. Eberhardt (Speech Processing Laboratory, Department of Electrical and Computer Engineering, Johns Hopkins University, Baltimore, MD 21218), and Marilyn E. Demorest (Department of Psychology, University of Maryland Baltimore County, Catonsville, MD 21228)

A study evaluated several transformations of voice fundamental frequency (F_0) for use by a single-channel vibrotactile device to supplement lipreading. The experimental procedure involved lipreading stimuli from a corpus of over 1500 sentences stored on a video laserdisk [L. E. Bernstein and S. P. Eberhardt, 1986, Johns Hopkins Lipreading Corpus]. There were five experimental conditions. Three transformations of voice F_0 were tested: two transformed F_0 to a pulse rate code, the third employed two sine waves that were amplitude modulated as a function of F_0 . Condition 4 provided both F_0 and a second stimulator indicating high-frequency speech energy. Condition 5 was visual alone. Subjects were 15 normal-hearing adults in a multiple single-subject design with alternating baseline and test sessions. Analyses of covariance for words correct, words per stimulus sentence, and syllables per sentence showed that all subjects improved in lipreading. Subjects who received a single vibrator for voice F_0 lipread more successfully than subjects who did not receive the vibrotactile supplement or who received the dual tactile stimulation.

J4. Tactile speech reception using augmented Tadoma. C. M. Reed, W. M. Rabinowitz, and N. I. Durlach (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Results obtained with Tadoma have set a new standard for the tactile communication of speech [Reed *et al.*, *J. Acoust. Soc. Am.* **77**, 247–257 (1985)]. These results, however, are inferior to those obtained in the normal auditory domain: they are roughly comparable to those obtained by normal-hearing subjects listening to speech in signal-to-noise ratios of 0–6 dB. The goal of the current research is to augment Tadoma with supplementary tactile displays to improve the perception of speech segments. Among the supplementary displays investigated were a vibrotactile display of tongue contact with the hard palate and a vibrotactile display of the short-term speech spectrum. The tongue-palate-contact display was effective for improving the discriminability of pairs of consonants difficult to distinguish through Tadoma; however, neither this display nor the multichannel spectral display proved to be effective for augmenting vowel discriminability. A third approach to augmenting Tadoma currently under investigation involves the tactile reception of cued speech. [Work supported by NSF.]

3:05

J5. Temporal and spatiotemporal tactile transforms of voice fundamental as an aid to lipreading. Arthur Boothroyd, Theresa Hnath-Chisolm, and Laurie Hanin (Graduate School, City University of New York, 33 W. 42 Street, New York, NY 10036)

Lipreading scores of naive subjects rise dramatically when the visible speech movements are supplemented by auditory presentation of voice fundamental frequency (F_0). If it can be shown that the tactual transmission of F_0 provides similar results, then there is the possibility of developing an effective lipreading aid for deaf subjects. We have tested two vibrotactile transforms of F_0 . One is temporal. It involves single-channel stimulation with a constant-amplitude square wave, of frequency $F_0/2$, at a fixed locus. The second is spatiotemporal. It uses the same stimulus, but locus varies, displacement being proportional to $\log(F_0)$. This transform requires a multichannel stimulator. In psychophysical experiments, the temporal display has been found to provide roughly 1/3-oct resolution at the fingertip, but over 1 oct on the forearm. With appropriate scaling, an eight-channel spatiotemporal display provides approximately 1/3-oct resolution on the forearm. With 16 channels, 1/6 of an octave resolution is possible. Experiments with speech stimuli have shown that both displays enhance the visual perception of phonetic contrasts that involve durational cues, but that the multichannel display is a more effective lipreading supplement for contrasts involving intonation contrasts. When sentence material was used with naive subjects, the two displays provided only a small supplement to lipreading. In one experienced lipreader, however, working with the multichannel display over a 13-week period, supplemented performance was significantly better than unsupplemented performance. The addition of the spatiotemporal transform of F_0 to the visual signal reduced the number of word recognition errors by 50%. A wearable version of the multichannel display has been developed and is currently in use in field studies. [Work supported by NIH PPG#17764.]

3:35–3:50

DISCUSSANT: D. Kimbrough Oller, *University of Miami*

3:50–4:05
Break

Contributed Papers

4:05

J6. A wearable multichannel vibrotactile aid for the deaf. A. Maynard Engebretson, Fengmin Gong, and Michael P. O'Connell (Central Institute for the Deaf, 818 S. Euclid Avenue, St. Louis, MO 63110)

It has been determined that hearing-impaired and normal-hearing subjects can learn to identify words using only a tactile vocoder [Brooks and Frost, *J. Acoust. Soc. Am.* **74**, 34–39 (1983)]. Further studies have shown that certain multichannel tactile vocoders provide significant help in understanding connected discourse as an aid in lipreading [Weisenberger and Miller, *J. Acoust. Soc. Am.* **82**, 906–916 (1987)]. A review of this research is given in a companion paper (Weisenberger, Abstract J1). Motivated by these results, a wearable, 16-channel vibrotactile aid has been developed that can be used by hearing-impaired subjects during normal daily activities. The system consists of a body-worn processor that is $3.5 \times 5.5 \times 1.5$ in. in size and a vibrator array worn on the arm. In addition

to the 16-channel vocoder algorithm, the microprocessor-based system can be programmed to implement other processing algorithms using the existing hardware. The system and its performance characteristics will be described. [Work supported in part by NIH.]

4:17

J7. Evaluation of a multichannel electrotactile device for the hearing impaired—A case study. Linda Kozma-Spytek and Janet M. Weisenberger (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

A multichannel electrotactile device for the hearing impaired was evaluated as an aid for speech reception and production with a 12-year-old profoundly hearing-impaired child enrolled in an oral school for the deaf. Over an 8-month period the subject received training in speech reception

skills in a hierarchy of tasks ranging from discrimination of minimal paired words to connected discourse tracking. Aspects of the subject's speech production skills were evaluated by obtaining ratings from a group of teachers of the hearing impaired, who viewed a videotape of the child during syllable production and connected discourse tracking with and without the device. Results of speech reception testing indicated that the device allowed good discrimination of minimal paired words based on manner contrasts, but poor discrimination based on place contrasts. During tracking, lipreading-pulse-tactile aid conditions were superior to lipreading alone. Results of the speech production evaluation showed that syllable identification was better and ratings on several aspects of speech production were higher under tactile aid conditions. [Work supported by NIH.]

4:29

J8. Tactile reception of fingerspelling and sign language. C. M. Reed, L. A. Delhorne, and N. I. Durlach (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Previous work on tactile speech reception by deaf-blind individuals has focused on the Tadoma method [Reed *et al.*, *J. Acoust. Soc. Am.* **77**, 247-257 (1985)]. Two additional methods of tactile communication, both of which are adaptations of methods designed for the visual sense but are used extensively within the deaf-blind community, are tactile fingerspelling and tactile signing. The goal of the current research was to document the communication abilities of highly experienced deaf-blind users of tactile fingerspelling and tactile signing. Experiments were conducted to determine reception accuracy for sentence-level materials as a function of rate of presentation for each of the two methods. The results of these experiments will be compared to those obtained for visual reception of fingerspelling and signing as well as to results obtained through the Tadoma method of tactile speech communication. [Work supported by NIH.]

4:41

J9. Stress contrast perception by the hearing impaired: Auditory, tactile, auditory-tactile. Janet Reath Schoepflin and Nancy S. McGarr (Speech and Hearing Sciences, Graduate School and University Center, City University of New York, New York, NY 10036)

The perception of stress contrasts in deaf subjects was assessed under three conditions—auditory only, tactile only, and combined auditory-tactile. Test stimuli were three disyllables generated in a two-stage process: multiple repetitions of each disyllable were produced by a normally hearing speaker using normal or exaggerated stress on either the first or second syllable; the averaged acoustic values for peak amplitude, vowel duration, and peak fundamental frequency for each disyllable were then resynthesized and manipulated to create a stimulus set containing none, one, two, and three of the acoustic cues denoting stress. Subjects were asked to indicate whether stress occurred on the first or second syllable for each stimulus item. Performance was above the level of chance in all three conditions. In the exaggerated stress production mode, performance in

the tactile condition exceeded performance in the auditory condition for those stimulus items containing a frequency cue. [Work supported by #NS-17764.]

4:53

J10. Reliability of individual differences in lipreading. Marilyn E. Demorest (Department of Psychology, University of Maryland Baltimore County, Catonsville, MD 21228), Lynne E. Bernstein, and Silvio P. Eberhardt (Speech Processing Laboratory, Department of Electrical and Computer Engineering, Johns Hopkins University, Baltimore, MD 21218)

Evaluation of the benefits of sensory aids requires stimulus materials whose psychometric characteristics are known. Generalizability theory was applied to data from two experiments to estimate measurement error arising from different materials, different talkers, and practice. Stimulus materials consisted of CID sentences and CV nonsense syllables produced by a male and a female talker and stored on video laserdisk. In experiment I, 104 normal-hearing subjects lipread the CID sentences in a single test session. Results indicate that there are substantial individual differences in lipreading performance among subjects, but that there are also systematic differences among sentences and among talkers that must be taken into account in interpreting test performance. In experiment II, 15 normal-hearing subjects who participated in an intensive training protocol involving vibrotactile supplements to lipreading were given pre- and posttests on the CID sentences and on the CV syllables. Individual differences in performance on sentences were highly stable over the training period, suggesting relatively uniform improvements due to training. Reliability over time was lower for the CV syllables and correlations between the CV and sentence materials were weak both at the pretest and posttest. [Work supported by NIH.]

5:05

J11. The use of tactile aids with deaf-blind children. Barbara Franklin (Department of Special Education, San Francisco State University, San Francisco, CA 94132)

This paper will present the results of the first year of a 3-year study designed to compare the effects of a 2-channel (Tactaid II) and a 16-channel (Tacticon) aid on expressive and receptive communication skills of deaf-blind children. The Tactaid vibrators are worn on the wrist and the Tacticon is worn as a belt of electrical stimulators on the abdomen. Six children, ranging in age from 22 months to 18 years, from the San Francisco Bay area are participating in the study. A total of three communicative behaviors will be selected for each child (e.g., number of purposeful vocalizations). A single-subject design will be used to compare their communication behaviors in three experimental conditions—no device (control), 2-channel device, and 16-channel device. Only one behavior will be observed at a time, resulting in three separate sub-studies per child. This paper will present the results of Sub-Study #1. Despite the tactile defensiveness often exhibited by deaf-blind children, the participants have been tolerating both devices. [This research is being supported by U.S. Dept. of Educ. Grant #G008630416.]

Session K. Psychological and Physiological Acoustics II: Loudness and Auditory Fatigue

Søren Buus, Chairman

Department of Electrical and Computer Engineering, Northeastern University, 360 Huntington Avenue, Boston, Massachusetts 02115

Contributed Papers

1:15

K1. Absolute magnitude estimation of loudness by adults and children. George A. Gescheider and Amy A. Collins (Department of Psychology, Hamilton College, Clinton, NY 13323)

Twenty-three subjects performed absolute magnitude estimation of the lengths of a series of lines and the loudness of a series of tones as well as cross-modality matching between loudness and perceived line length. The results support the hypothesis that subjects judge stimuli on an absolute scale. Specifically, for 9 out of 12 adults and 9 out of 11 children, lines and tones assigned the same number in magnitude estimation were judged to be subjectively equal in cross-modality matching. A correction procedure was employed to eliminate the effects of idiosyncratic number usage from the magnitude estimations of loudness. This correction procedure, consisting of dividing the loudness exponent by the line length exponent, produced power function exponents for loudness that were virtually identical for magnitude estimation and cross-modality matching. Implications of the results for clinical measurement of loudness are discussed.

1:30

K2. Deviations from a power-function near miss and their relation to loudness functions. William S. Hellman (Department of Physics, Boston University, Boston, MA 02215) and Rhona P. Hellman (Department of Psychology, Northeastern University, Boston, MA 02115)

In a previous paper [W. S. Hellman and R. P. Hellman, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S53-S54 (1987)], the neural-count function $N(I)$ was derived from the intensity-jnd function $J(I)$ through the relation $N(I)^{1/2} = (h/2) \int dI / [J(I)] + a$. Loudness functions were then generated by the prescription $L = k[N(I) - N]$. Using a power-function near miss for $J(I)$, good agreement between the measured and calculated loudness values for pure tones was obtained. While a power function yields a simple evaluation of the integral for $N(I)^{1/2}$, the results of many recent jnd studies do not easily conform to a power-function fit. In order to determine how the departures from a power-function near miss affect the form of the loudness function within the model, the integration was performed over the segmented intensity jnd functions observed for gated and continuous tones. The calculated loudness functions and their respective input intensity jnd functions are shown for frequencies of 250 and 1000 Hz. In spite of the segmented, and in some instances, nonmonotonic intensity jnd functions, the results reveal that the smoothing action of the integration over $J(I)$ produces loudness functions consistent with experimental results. [Partially supported by the Rehabilitation Research and Development Service of the VA.]

1:45

K3. Overall loudness of tone-noise complexes: Measured and calculated. Rhona Hellman (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115) and Eberhard Zwicker (Institute of Electroacoustics, Technical University of Munich, Munich, Federal Republic of Germany)

Loudness measured for single-tone-noise complexes [R. P. Hellman, *J. Acoust. Soc. Am.* **72**, 62-73 (1982)] is compared to loudness calculat-

ed in accordance with a loudness-calculation program based on ISO 532 B [E. Zwicker, H. Fast, and C. Dallmayr, *Acustica* **55**, 63-67 (1984)]. Data are given for single tones centered within the spectrum of broadband-flat, low-pass, and high-pass noises. The measured and calculated loudness functions exhibit a similar pattern of loudness growth. Both measured and calculated loudness of the tone-noise complexes are non-monotonic functions of the overall SPL of the complex. Thus two tone-noise combinations at nearly the same overall SPL can produce markedly different loudness values. These results hold over a 30-dB range from about 73-103 dB. The results also show that the magnitude of loudness depends on the spectral shape of the noise and the frequency of the added tone. The close agreement between the measured and calculated loudness-growth patterns means that basic psychoacoustical principles governing loudness and masking can account for the observed effects. [Supported by the Rehabilitation Research and Development Service of the VA and by the Deutsche Forschungsgemeinschaft.]

2:00

K4. Psychometric functions for level discrimination. Søren Buus (Communication and Digital Signal Processing Laboratory, 409 DA, Northeastern University, Boston, MA 02115), Christine R. Mason, and Mary Florentine (Communication Research Laboratory, 133 FR, Northeastern University, Boston, MA 02115)

To determine the form of psychometric functions for 2I, 2AFC level discrimination, ten increment levels were presented in random order within blocks of 100 trials. Stimuli were chosen to encompass a wide range of conditions and difference limens: eight 10-ms tones had frequencies of 0.25, 1, 8, or 14 kHz and levels of 30, 60, or 90 dB SPL; two 500-ms stimuli also were tested: a 1-kHz tone at 90 dB SPL and a white noise at 63 dB SPL. For each condition, at least 20 blocks were presented in mixed order. Results for three normal listeners show that the sensitivity d' is nearly proportional to $\Delta L \{ = 20 \log[(p + \Delta p)/p] \}$, where p is pressure over the entire range of difference limens. When d' is plotted against Weber fractions for pressure $\Delta p/p$ or intensity $\Delta I/I$ exponents of the best-fitting power functions decrease with increasing difference limens and are less than one for large difference limens. These results indicate that the transformation from stimulus intensity to decision variable may be approximately logarithmic and that ΔL —plotted on a logarithmic scale—is an appropriate representation of level discrimination performance. [Work supported by NIH.]

2:15

K5. Investigation of a model of loudness. P. S. Chien¹⁾ (Center for Research in Speech and Hearing Sciences, City University of New York, New York, NY 10036) and J. L. Hall (AT&T Bell Laboratories, Room 2D526, 600 Mountain Avenue, Murray Hill, NJ 07974)

A model of loudness summation by Schroeder *et al.* [*J. Acoust. Soc. Am.* **66**, 1647-1652 (1979)], which was used by them to provide an objective quality measure of digital speech coding, is being investigated. (1) Iso-loudness contours generated by the model are compared to iso-loudness contours generated by human subjects. (2) The spreading function

used in the model to calculate excitation levels is converted from the critical-band domain to the frequency domain for various center frequencies. The resulting curves are compared to tuning curves measured in the cat. (3) The effect of partial masking on the relative loudness of pairs of pure tones is examined. (4) The model is used to predict the loudness of broad bands of noise from the loudness of narrow bands of noise centered at several frequencies. Studies (1)–(4) have revealed strengths and shortcomings of the model. ^{a)} Work done as a consultant to AT&T Bell Laboratories.

2:30

K6. Loudness adaptation in musicians and nonmusicians. C. Baruch and M.-C. Botte (Laboratoire de Psychologie Expérimentale, CNRS, 28 rue Serpente, 75006 Paris, France)

Measurements of simple and induced loudness adaptation were made on 10–12 musicians and 10–12 nonmusicians by the method of successive magnitude estimations. Simple adaptation (SA) was measured for a 500-, 1000-, or 4000-Hz tone presented monaurally for 3 min at 10 dB SL. Induced adaptation was measured for a continuous 1000-Hz, 60-dB SPL right-ear tone; loudness was judged before, during, and after each occurrence of an intermittent inducer tone set to five frequencies from 500–3260 Hz and presented every 30 s for 10 s to the same ear (ipsilaterally induced adaptation, IIA) or to the left ear (contralaterally induced adaptation, CIA). We also measured CIA with a rapid intermittency of every 1 s for 0.5 s. The level of the inducer was 60 dB for CIA and 75 dB for IIA. Musicians showed less adaptation than nonmusicians whatever the type of adaptation: For the musicians, the maximum loudness decrease was 44% under SA, 30% under CIA, 30% under IIA; for the nonmusicians, this maximum was 70% under SA, 70% under CIA, 41% under IIA, but differences were statistically significant only for SA and for CIA with the rapid intermittency. Moreover, musicians showed significant adaptation over a less extended range of frequencies than nonmusicians.

2:45

K7. A pure-tone-induced temporary threshold shift on normal hearing and noise-induced permanent threshold shift. I. M. Young and L. D. Lowry (Department of Otolaryngology, Jefferson Medical College of Thomas Jefferson University, Philadelphia, PA 19107)

Temporary threshold shift after an exposure to a pure tone was measured by automatic audiometry on three subjects with normal hearing and three subjects with noise-induced permanent threshold shift. Subjects were exposed to a stimulating pure tone with a frequency 1500 Hz through an earphone at an intensity of 110 dB SPL for a duration of 10 min. Temporary threshold shift was measured beginning at 5 s after the cessation of the stimulation. In normal hearing subjects, the greatest shift was observed at the frequency area of 2000 Hz as demonstrated previously. In subjects with noise-induced permanent threshold shift, temporary threshold shift was demonstrated not only at the frequency region of 2000 Hz, but also at higher frequencies with permanent threshold shift. Post-exposure effects at higher frequencies showed the greater threshold shift for the steady tone resulting in increased separation between normal amplitude pulsed tone tracings and markedly reduced amplitude steady tone tracings. Similar post-exposure findings were observed on a subject with congenital nonprogressive bilateral symmetrical sensorineural loss of high frequencies similar to noise-induced permanent threshold shift. Spread of temporary threshold shift at the higher frequencies was discussed and compared with the greater spread of masking effect at the higher frequencies.

3:00

K8. Auditory fatigue and induced loudness adaptation. S. Charron, M.-C. Botte (Laboratoire de Psychologie Expérimentale, CNRS, 28 rue

Serpente, 75006 Paris, France), S. Mönikheim (Institut für Psychologie, Würzburg, Austria), and B. Scharf (Northeastern University, Boston, MA 02115)

Measurements of induced loudness adaptation and temporary threshold shift (TTS) were made on 48 young subjects. Loudness adaptation of a continuous 60-dB test tone was induced in the right ear by an intermittent 1000-Hz inducer tone at 90 dB, presented every 30 s for 20 s. The loudness of a 1000-Hz or 1160-Hz test tone at 60 dB was measured after each occurrence of the inducer by the method of successive magnitude estimations. Induced adaptation caused the loudness of the continuous tone to decrease on the average by 38% (the equivalent of 14 dB) after 120 s. In a separate session, the subject's right ear was exposed for 45 min to a 1000-Hz tone at 90 dB SPL. One minute after exposure, thresholds were measured by Békésy tracking for 4 min. The maximum TTS, averaged across subjects, was 20.4 dB at a mean frequency of 1635 Hz. The correlation between maximum TTS and the amount of induced adaptation was 0.83. Thus ipsilaterally induced adaptation (IIA), which is akin to temporary loudness shift, may stem from cochlear mechanisms just as TTS does. Also, IIA could become the basis for an audiological test to identify those individuals most susceptible to auditory fatigue. [Work supported by Ministère de l'Environnement and NIH.]

3:15

K9. Reduction of auditory damage due to intermittence. W. Dixon Ward (2630 University Avenue S. E., Minneapolis, MN 55414)

Two groups of chinchillas were given daily exposures, 5 days/week for 9 weeks, to 700–2800-Hz noise at 110 dB SPL. Each day, one group received a single 48-min exposure, the other group a series of 40 1.2-min bursts spaced at 12-min intervals. Whereas the continuous single exposures produced a median permanent threshold shift (PTS) of 16 dB at 1, 2, and 4 kHz and destruction of 33% of the outer hair cells (OHCs), the intermittent exposures resulted in a PTS of only 6 dB and less than 5% destroyed OHCs. Inasmuch as the latter group suffered about the same cochlear damage as (and less PTS than) a group given 45 daily 48-min exposures at 102 dB SPL, one can infer that an 8-dB "correction for intermittence" of a 110-dB exposure is approximately correct, in accordance with present OSHA regulations governing industrial noise exposure. [Research supported by NIH Grant 12125.]

3:30

K10. Threshold recovery functions following impulse noise trauma. Roger P. Hamernik, William A. Ahroon (Auditory Research Laboratories, State University of New York, Plattsburgh, NY 12901), and James H. Patterson (Sensory Research Division, U.S. Army Aeromedical Research Laboratory, Fort Rucker, AL 36362-5292)

An analysis of the pure-tone threshold recovery functions obtained from 118 chinchillas exposed to high-level impulse noise showed that there are at least three distinctly different types of threshold recovery functions. Type I: the classical recovery function that declines monotonically with increasing postexposure time; type II: a delayed recovery, i.e., for a period as long as 6 h following removal from noise, the pure-tone threshold remains elevated and stable before a monotonically declining recovery process sets in; and type III: the growth function, i.e., over a period of at least 6 h following removal from the noise, pure-tone thresholds continue to get worse before a stable monotonically declining recovery process begins. Frequencies characterized by a type III recovery process show more PTS and sensory cell loss than do those characterized by the type I recovery process. The existence of three distinctly different audiometric recovery functions implies the existence of different mechanisms of cochlear trauma.

Session L. Noise II: Analysis and Reduction of Noise

Krish Krishnappa, Chairman

Engineering Laboratory M-7, National Research Council of Canada, Montreal Road, Ottawa, Ontario K1A 0R6, Canada

Chairman's Introduction—1:30

Contributed Papers

1:35

L1. Tone generation by impingement on plates by supersonic jets at various over-pressures. Alan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77004)

Sonic jets at "moderate" over-pressure impinging on normal, flat, rigid plates generate intense *principal* tones by two distinct mechanisms [see *J. Acoust. Soc. Am. Suppl. 1* 81, S96 (1987)]. For small plates $d/D < 2$, (d = plate diam, D = nozzle diam) a local "feedback" occurs in the subsonic region between the stand-off shock wave and the plate, akin to that in the Hartmann air-jet generator when operated at the same pressure, with $\lambda/D \approx 4$ (λ = acoustic wavelength). For large plates ($d/D > 2$) an edgetone-like acoustic feedback to the nozzle occurs through the ambient atmosphere external to the jet and $\lambda/D \approx 2$. The unstable shock wave is excited from the downstream and upstream sides, respectively. *Secondary* tones also arise, occurring *instead of* the principal tone. Some have $\lambda/D \approx 1$, and are directly related to the large plate tones, even for small plates, while others are also closely related to the edgetone-like tones, but are of slightly longer wavelength. This classification enables a rational analysis and interpretation to be made of data in the sparse literature, including those for supersonic jets and jets at "high" over-pressures, and "very high" over-pressures, in which case the tone mechanisms appear to merge, with $\lambda/D > 4$.

1:50

L2. Flow/acoustic interaction in duct inflow. Margaret C. Quinn (The Institute of Sound and Vibration Research, Southampton University, Southampton SO9 5NH, United Kingdom)

Sound generation in the convection of turbulence into a flow intake duct is discussed by examination of an idealized problem. Ideal fluid is in motion with uniform low, subsonic velocity above and parallel to a plane rigid wall. A thin, rigid, semi-infinite plate is parallel to the wall and the fluid flows past its leading edge. The radiated sound is calculated for a turbulent eddy modeled by a weak line vortex that is allowed to convect passively past the leading edge of the plate. Account is taken of the contribution to the radiated sound from the disturbance produced by the convecting vortex in the boundary layers on each side of the plate by means of Howe's (1981) theory of displacement thickness fluctuations (*Proc. R. Soc. London Ser. A* 374, 543-568), the strength of such disturbances being fixed by a leading edge Kutta condition. It is concluded that the predicted level of the radiated sound is substantially reduced by these boundary layer disturbances relative to when they are neglected. The case of the convection of a frozen two-dimensional gust is also considered. Examination of the analogous problem of plane-wave radiation from the duct shows that the presence of displacement waves enhanced both the far-field intensity in the ambient fluid and the reflected field within the duct. [Work supported by ARE Teddington, UK.]

2:05

L3. Further studies on panel absorbers. B. S. Sridhara and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

Panel absorbers are very useful in reducing the noise propagated in pipes and ducts in applications where reactive mufflers cannot be used. A panel absorber is made of a rectangular panel backed by an air gap. It is believed that this is the first time detailed studies have been made on the applications of panel absorbers in pipes and ducts. Governing differential equations are derived considering the lumped and distributed spring mass systems. The ordinary and partial differential equations are solved and the expressions for the natural frequencies and the admittance of the panel absorber are obtained. Equations for the insertion loss and the transmission loss of the panel absorber are derived. Ten different types of aluminum panel absorbers have been designed, fabricated, and tested in the Sound and Vibration Laboratory at Auburn University. Panel absorber parameters such as panel thickness, panel edge support (boundary) conditions, panel absorber mounting, and panel dimensions were considered during the experiments. The location of the panel absorber along the length of the pipe was also examined. The experimental results are very encouraging. Insertion loss of the first panel absorber was 2.8 dB, whereas the insertion loss of the latest panel absorber with all the improvements is 18 dB.

2:20

L4. Resonance conditions in branching structures. David T. Raphael and M. A. F. Epstein (Department of Anesthesiology, University of Connecticut Health Center, Farmington, CT 06032)

In complex branching structures, recursive algorithms are often employed to calculate impedances at successive nodes. For a rigid symmetric N th-order branching network terminating in 2^N closed branches, a technique is presented whereby the open-ended input impedance, and its corresponding resonance condition, can be explicitly formulated in terms of the branch lengths and cross-sectional areas, the angular frequency, and the integral harmonic number. In the low-frequency range, the resonance condition can be modified so as to facilitate volume estimation of the branching structures. Experimental studies with rigid second-order symmetric and asymmetric branching structures are discussed. [Work supported by the University of Connecticut Research Foundation.]

2:35

L5. Surface roughness effects on flow noise. George H. Christoph (Martin Marietta Laboratories, 1450 S. Rolling Road, Baltimore, MD 21227)

The Corcos wall pressure model was modified to incorporate surface roughness effects. The longitudinal cross-spectral density part of the Corcos model was modeled from rough wall data taken from the literature. The wall pressure frequency spectrum was nondimensionalized by the wall shear stress and boundary layer displacement thickness and shown to fit both smooth and rough wall data. The boundary layer parameters were obtained from a finite-difference boundary layer code developed by the author to account for surface roughness effects by the discrete roughness approach [G. H. Christoph and R. H. Pletcher, *AIAA J.* 21, 509-515 (1983)]. The blockage of the roughness height, shape and spacing is explicitly accounted for in this approach. Flow noise was calculated for a variety of roughness element heights and spacings in air and in water. The

calculations clearly show that element spacing as well as height is important and that the viscous sublayer height, not the boundary layer thickness, determines the influence of roughness on flow noise. Thus roughness

is typically more important in water than in air because of the thinner viscous sublayer. Also, the effect of transducer size and viscoelastic layer thickness are shown for rough surfaces.

**2:50-3:00
Break**

3:00

L6. Helicopter rotor speed effects on farfield acoustic levels. Arnold W. Mueller, Otis S. Childress, Jr. (NASA Langley Research Center, M.S. 460, Hampton, VA 23665), and Mark Hardesty (McDonnell Douglas Helicopter Company, Mesa, AZ 85201)

The design of a helicopter is based on an understanding of many parameters and their interactions. For example, in the design stage of a helicopter, the weight, engine, and rotor speed must be considered along with the rotor geometry when considering helicopter operations. However, the relationship between the noise radiated from the helicopter and these parameters is not well understood, with only a limited set of model and full-scale field test data to study. In general, these data have shown that reduced rotor speeds result in reduced farfield noise levels. This paper will review the status of a recent helicopter noise research project designed to provide flight experimental data to be used for further understanding of helicopter rotor speed effects on farfield acoustic levels. Preliminary results will be presented relative to tests conducted with a McDonnell Douglas Helicopter Company model 500E helicopter operating with the rotor speed as the control variable over the range from 103% N2 to 75% N2 and the forward speed maintained at a constant value of 80 knots.

3:15

L7. Fan noise and unsteady rotor forces. Wen-Shyang Chiu, Gerald C. Lauchle, and D. E. Thompson (Graduate Program in Acoustics, Applied Research Lab, Pennsylvania State University, Box 30, State College, PA 16804)

The noise radiated by a subsonic, axial-flow fan at its rotational frequency and harmonics is related to the nonsteady force field created at the rotor blade/fluid interface. This force field is highly dependent on the time-invariant flow distortions that enter the fan. In this basic study, a typical cooling fan used in the electronic and computer industry was instrumented with an unsteady axial force sensor. Its output is proportional to the total unsteady axial force created by the rotor. The inflow field of the fan was systematically distorted by placing a small cylinder at various positions in the inlet plane. The nonuniform, three-dimensional flow field entering the rotor was measured by traversing a miniature five-hole pressure probe. The total pressure outputs from this probe can be related to the axial, tangential, and radial velocity vectors. Fourier decomposition of the inflow velocity data is coupled with analysis to give information on the unsteady rotor force harmonic content. The on-axis sound pressure levels were measured and compared to coherent output power spectra involving the unsteady force sensor and the microphone. Very good coherence at the discrete tones is observed. [Work supported by IBM Corp.]

3:30

L8. Model and full-scale testing of a modified cutoff bar for a centrifugal fan. Frank H. Brittain (Bechtel National, Inc., San Francisco, CA 94105) and David A. Nordstrand (Detroit Edison Co., Detroit, MI 48226)

Controlling centrifugal fan noise by modifying the design of the fan is very appealing because the cost could be considerably lower than for an

in-duct silencer. Pressure drop, fouling, and other problems frequently associated with silencers can also be avoided. Previous attempts at modifying an existing centrifugal fan to control noise have often met with limited success. To reduce the cost of controlling blade passage tones in the community, modification of the induced draft (ID) fans at a power plant was evaluated. Model testing was conducted for ten different designs of the cutoff bar for the centrifugal fan. The noise reductions and fan efficiencies varied widely for six of the ten cutoff bars tested. The best design gave better noise reductions than expected and higher efficiencies at usual settings of the variable inlet vanes. The best cutoff bar design was installed (full scale) on one of the ID fans and fields tested. The noise reductions on the full-scale fan tests were better than obtained for the scale model tests. Results of both the model and full-scale tests are presented, compared, and discussed.

3:45

L9. Frequency domain analysis of dipole source strength and correlation area. David M. Yeager (IBM Acoustics Laboratory, P. O. Box 390, C18/704, Poughkeepsie, NY 12602)

The generation and radiation of noise from a rigid surface immersed in a turbulent flow field may be characterized by the strength of the dipole-like sources associated with the normal surface pressures acting on the fluid at the surface, and by the regions over which these sources are partially correlated. New frequency-domain expressions have been developed for the dipole source strength per unit area, based on mean-square pressure and acoustic intensity, and for the correlation area. The equations are similar to time-domain expressions developed earlier [T. E. Sidon, *J. Acoust. Soc. Am.* **53**, 619-633 (1973)]. The equations show that the correlation area can be thought of as a complex radiation efficiency that is normalized by its maximum value, the total area of the surface immersed in turbulence. Both the correlation area and dipole source strength per unit area reflect the noise generation characteristics of the local turbulence regions and mutual interactions between regions. Results of experiments will be presented in which a single blade, removed from a rotating centrifugal air moving device, was immersed in a turbulent flow field.

4:00

L10. Scale model of an active noise reduction system. O. L. Angevine (Angevine Acoustical Consultants, Inc., P.O. Box 725, East Aurora, NY 14052)

Scale modeling has been used often for predicting acoustical properties of architectural structures such as music halls. In this experiment, it is an active noise attenuating system that is modeled. By using a scale factor of 8 or 16, tests can be conducted on a table top simulating those of a full-scale system at a distance of 24 m. Since active attenuation is most successful at frequencies below about 500 Hz, the scale frequencies remain in the audible spectrum, and measurement does not require special equipment. [Work supported by EPRI and NMPIC.]

Session M. Underwater Acoustics II: Arctic Acoustics II (Précis-Poster Session)

Robert Francois, Chairman

Applied Physics Laboratory, University of Washington, Seattle, Washington 98195

Contributed Papers

Following presentation of the précis, posters will be on display until 5:00 p.m.

1:30

M1. Ambient noise in Baffin Bay. Joseph B. Farrell and Garry J. Heard (Defence Research Establishment Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

This paper presents the results of ambient noise measurements taken in Baffin Bay in August–September of 1986. Noise measurements were taken in conditions ranging from open water to iceberg-infested water near the edge of pack ice. Differences in the character of ambient noise at the experimental sites will be noted and discussed. The frequency range investigated in this work was 10–1000 Hz. Comparisons will be made with noise measurements taken by Defence Research Establishment Atlantic (DREA) in Baffin Bay in the 1970s.

1:34

M2. Super-resonant scatter under ice sheets and in full spaces. Ivan Tolstoy (Knockvennie, Castle Douglas, SW Scotland) and Alexandra Tolstoy (Naval Research Laboratory, Washington, DC 20375)

Systems of monopole scatterers interacting through multiple scatter may develop normal modes, i.e., true resonant behavior at certain frequencies (the equivalent source coefficients of each scatterer have real poles). This occurs when the single scatter cross sections are large enough to give significant interaction. This happens for highly tuned scatterers, e.g., gas-filled balloons, bubbles, and thin shells in water, insonified at frequencies near their characteristic “bubble” frequencies. This kind of *super-resonance* (SR) occurs when the system is at or near the interface with an elastic medium, e.g., a thin elastic plate [I. Tolstoy, *J. Acoust. Soc. Am.* **80**, 282–294 (1986)]. Energy transport between scatterers is then mediated by *boundary modes* excited by the scatterers acting as point sources (e.g., the flexure waves of a thin plate). The case of two and three bubble systems at an ice–water or lucite–water interface is explored, with particular reference to the parameters ka , kl (k is the acoustic wavenumber, a the scatterer radius, l the spacing between scatterers) and the plate elastic constants. For a pair (doublet) the pressure amplification re : the free-field value is $\approx 2 \times 10^3$ for kl values as small as 0.5, with a directionality factor of 30: This is a passive, highly tuned, high magnification, strongly directional compact array ($\approx \lambda / 12$ size). A 30-Hz system would consist of two ballons of 10-cm radius near an ice boundary, with a discrete set of permissible l values ranging from about 5–20 m. Optimum spacings and magnifications depend upon the ice thickness (≈ 5 m). In an acoustic *full space*, energy exchange between scatterers is mediated less effectively by volume waves, and the scatter coefficient poles vanish: *Bona-fide* SR does not take place. Nevertheless, strong peaks may occur, giving magnifications $\approx 10^3$. This *quasiresonant* behavior is discussed for simple two-, three-, four-, and six-element plane and polyhedral configurations. However, true SR behavior is reintroduced in full spaces by partially blocking selected pathways between scatterers. [Work supported by ONR.]

1:38

M3. A study of reflection and refraction of waves at the interface of water and sea ice. C. H. Yew and X. Weng (Department of Aerospace Engineering and Engineering Mechanics, University of Texas at Austin, Austin, TX 78712)

The sea ice is modeled as a transversely isotropic brine-saturated porous medium, in accordance with the model proposed by Schwarz and Weeks (1977). The reflection and refraction of waves at the water–ice interface are studied. The numerical results obtained in this study indicate that the porosity of the ice has a noticeable effect on the wave reflectivity. Its effect, however, becomes small when the grazing angle of the incident wave becomes small. The existence of a thin and porous ice skeleton layer on the ice medium further reduces the reflectivity. The reduction of reflectivity becomes more distinct as the frequency of the incident wave is high. Numerical results point out that, in addition to the porosity, the ice layer thickness is the most dominant property influencing the under ice wave reflectivity. The calculated results show a qualitative agreement with the *in situ* experimental results by Yang and Votaw (1981).

1:42

M4. A modal approach to Arctic noise modeling with implied vertical coherence. R. B. Williams (Naval Ocean Systems Center, San Diego, CA 92152)

Analytic formulas of the modal attenuation coefficients appropriate for deep Arctic under-ice conditions, and modal energy partitioning of ambient noise sources are proposed. These functions can be used, together with various assumed spatial and temporal distribution functions to calculate various statistical ambient noise functions, such as vertical coherence. Vertical noise data between 15 and 50 Hz have been analyzed for model comparison, using a bandwidth of about 5 Hz. These data show that the vertical coherence usually has a $\sin \pi/x$ dependence, with the first zero crossing at one wavelength. Sometimes, however, zero crossing lengths of more than two wavelengths are observed. Longer correlation lengths are consistent with much of the noise energy radiating from a small region, although other interpretations are possible. Initial simple model calculations show agreement with the data. A strong nearby source of noise will cause the correlation to lengthen, while several more distant sources support the one wavelength zero crossing result. Initial sensitivity calculations, however, show that the vertical coherence function by itself is not a good validator of the model functions. Other statistics will be sought to further examine model validity. [Work supported by NORDA and NOSC 6.2 Arctic programs.]

1:46

M5. Under-ice scattering mechanisms. R. H. Mellen (PSI Marine Sciences, New London, CT 06320) and P. S. Scheifele (Naval Underwater Systems Center, New London, CT 06320)

Underwater acoustic propagation in the Arctic Ocean is characterized by a refractive surface channel with a rough boundary. Scattering estimates based on free-surface theory have proven low by more than a factor of 2 in forward-scatter loss and low by 20 dB or more in backscatter strength. Failure to account for either attenuation or backscattering indicates that impedance effects must be involved. The ice layer is modeled as a uniform elastic solid. Perturbation analysis shows scattering from slopes as well as displacements. Displacements produce pistonlike reradiation in addition to the usual vertical-dipole type associated with free surfaces. However, the “rocking” horizontal dipole produced by slopes is evidently the dominant mechanism at the lower frequencies and estimates of both

attenuation and backscatter strength show reasonable agreement with experimental data when included. [Work supported by NUSC/NL.]

1:50

M6. Sound absorption in Arctic regions. R. H. Mellen (PSI Marine Sciences, New London, CT 06320), P. S. Scheifele, and D. G. Browning (Naval Underwater Systems Center, New London, CT 06320)

Sound absorption in seawater involves relaxations of magnesium sulfate, boric acid, and magnesium carbonate. Absorption varies with both region and depth, due mainly to the pH dependence of the boric acid relaxation. The nominal seawater pH range is 7.7–8.3 and magnitudes can vary over nearly a factor of 4 at low frequencies. A global model for World Ocean prediction has been developed, employing contour charts of the pH factor at several depths. Profiles for integration of loss along ray paths are generated by algorithm. The pH range in the Arctic is nominally 8.0–8.3 and similar contour charts are provided for the pH factor in these regions. Although long-range propagation is generally limited by under-ice scattering, absorption can be the dominant loss mechanism under ice-free conditions and for ranges less than 50 km at sonar frequencies with an ice cover. Predicted absorption spectra are presented and compared with the limited data available. [Work supported by NUSC/NL.]

1:54

M7. A bistatic, high-frequency, under-ice, acoustic scattering model. Garner C. Bishop (Bldg. 679, Code 8211, Naval Underwater Systems Center, Newport, RI 02841)

A model has been developed to evaluate the scatter produced by a high-frequency acoustic pulse originating from an arbitrarily located source, incident on the under-ice surface characteristic of pack ice regions of the interior Arctic and detected by an arbitrarily located receiver. Measured, two-dimensional, under-ice, acoustic profile data and several empirical results that relate various geometric parameters of the large scale under-ice relief features, e.g., ice keels are used to construct a three-dimensional, bimodal, under-ice surface model consisting of first-year ice keels and sloping flat ice regions. A first-year keel is modeled as an ensemble of randomly oriented ice blocks on a planar surface inclined at some slope angle with respect to a horizontal plane at sea level. Ice blocks are modeled as layered, viscoelastic, rectangular solids. A region of flat ice is modeled as a smooth planar surface whose slope angle is less than some arbitrary minimum keel slope angle. The Helmholtz–Kirchhoff integral and the Kirchhoff approximation are used to evaluate the scattered field. Time is partitioned into bins and then, beginning at the bin corresponding to the time the scattered field of each scattering facet arrives at the receiver, the scattered field is added coherently in all bins spanning the temporal duration of the incident pulse. The resulting scattered field time series is used to calculate the corresponding reverberation time series. Modeled reverberation levels are compared with measured levels for several Arctic sites and modest agreement is obtained.

1:58

M8. Observation of thermal ice-cracking in the Arctic Basin. P. Zakarauskas (Defense Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

It is now well known that thermal ice-cracking is a major contributor to under-ice ambient noise in certain regions of shore-fast ice during winter and spring [A. R. Milne, *J. Geophys. Res.* 77, 2177–2192 (1972)]. A report is given here of an observation of thermal ice cracking made by the Defense Research Establishment Pacific over polar pack ice during the spring of 1987. The number of events, as counted by a time-domain kurtosis method, is shown to follow the temperature changes of the air very closely. Thermal ice-cracking was the dominant ambient noise source during the periods of rapid air cooling. For thermal ice-cracking to occur, three conditions must be met: low temperature, low ice salinity, and little

or no snow cover. Since the only ice surfaces directly exposed to air in the vicinity of the experiment were the vertical sides of large blocks of ice in pressure ridges, it is hypothesized that the cracks could have originated there.

2:02

M9. An experimental study of the coherent under-ice reflectivity of sound in the Greenland Sea Marginal Ice Zone (MIZ). Patricia L. Gruber and Ronald L. Dicus (Code 5120, Naval Research Laboratory, Washington DC 20375)

The coherent component of acoustic under-ice reflectivity was investigated as a function of frequency and grazing angle in the MIZ. Measured reflectivity was compared with scattering theory predictions to test the hypothesis of ridge scattering dominance in the MIZ. Explosive source acoustic signals were received on a vertical array, during the Marginal Ice Zone Experiment (MIZEX 84), and deconvolved to separate the direct and ice-reflected arrivals. Ice-reflected signals received on individual hydrophones at different ranges and depths, and on different days, were aligned in time and coherently ensemble averaged within a moving 10-deg grazing angle window. The resulting reflectivity was found to decrease with increasing frequency from 64–256 Hz and with increasing grazing angle from 12–35 deg. Comparison was made with predictions from smooth elastic plate theory, perturbation theory for a rough pressure release surface, perturbation theory for a rough elastic surface, Twersky theory for hemispherical bosses on a plane, and Twersky theory for infinite, semielliptical cylinders on a plane. The Twersky theory for semielliptical cylinders gave a significantly better fit to the data than the other theories. This suggests that, although the under-ice topography may be a good deal more complicated, scattering may, nevertheless, be dominated by remnants of elongated pressure ridge keels. Remote sensing data, taken during the experiment, showed that the ice cover was primarily composed of small floes (< 200 m in diameter). Laser ice-surface height measurements were analyzed to obtain the pressure-ridge sail-height distribution. The Twersky theory modeling gave predictions within 1 dB of the measured mean reflectivity when a ridge keel-to-sail ratio of 6.5 was assumed.

2:06

M10. Attenuation of sound through laboratory grown saline ice. K. C. Jezek (U.S. Army Cold Regions Research and Engineering Laboratory, and Thayer School of Engineering, Dartmouth College, Hanover, NH 03755), T. K. Stanton (Department of Geology and Geophysics, University of Wisconsin, 1215 W. Dayton, Madison, WI 53706), and A. J. Gow (U.S.A. Cold Regions Research and Engineering Laboratory, Hanover, NH 03755)

Attenuation of acoustic waves transmitted through saline ice was studied *in situ* as ice formed in a large, outdoor pool. Amplitude of sonar pings (50–800 kHz) transmitted normal to the ice sheet surface were measured after passing through ice about 18 cm thick and having an average salinity of about 4 parts per thousand. Columnar crystals in the body of the ice sheet and a fine, dendritic structure near the base of the ice sheet (dendrites were about 0.5 mm thick) characterize the ice as being nearly identical to young, undeformed sea ice found naturally covering the polar oceans. Additional measurements were performed on newly grown saline ice about 3 cm thick. Measured attenuations through the 18-cm-thick ice ranged from about 20 dB at 50 kHz to 80 dB at 420 kHz. For the 3-cm-thick ice, total attenuation at 188 kHz was about 12 dB (compared to about 40 dB for the 18-cm-thick ice). That this represents an absorption of the acoustic wave by the ice is evidenced by two other observations: first, a low reflection coefficient measured on newly grown ice (about 0.04 at 188 kHz); second, measurements of scattered energy back into the transmitting region showed little redistribution of energy into nonspecular directions. Based on the observed decrease of attenuation per unit thickness of ice, significant absorption is attributed to the lower, porous and permeable portions of the ice. Presumably the important absorption mechanisms are associated with the pure ice dendrites and alternating planes of brine pockets.

M11. Elements of Arctic surface scatter: Part III, the head wave. Herman Medwin, Ken J. Reitzel,^{*)} and Michael J. Browne (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Laboratory scale model studies have the virtue of revealing the detailed behavior of the elements of scatter from a complex ice field. Previous model studies have demonstrated the quasidipole radiation pattern of low-frequency scatter from an open lead. It has also been shown that leads and ridges backscatter principally from a one wavelength extent near the least-time intercept of the radiation from a point source. At this time, evidence is presented that mode conversion generates a compressional wave that propagates parallel to the surface of the ice. This mode is driven, and will radiate into the water, at the critical angle. The reradiated sound, an example of a head wave, causes a significant precursor to appear before the specular reflection from the ice canopy. In agreement with theory, the measured head wave amplitude is proportional to $k^{-1} x^{-1/2} \times L^{-3/2}$ where k is the wavenumber in the water, x is the horizontal range, and L is the path length in the ice. [Work supported by the Office of Naval Research.] ^{*)}Ocean Acoustics Associates, 4021 Sunridge Rd., Pebble Beach, CA 93953.

2:14

M12. Laboratory scale investigation of the amplitude and phase of signals reflected from sea ice ridges. T. McLanahan (Naval Air Systems Command, Washington, DC 20361-5220), O. I. Diachok, and S. C. Wales (Naval Research Laboratory, Washington, DC 20375)

Acoustic reflectivity from plastic models of sea ice ridges on a thin plate were investigated for $0.67 < ka < 6$ (where k is the wavenumber and a is the mean ridge depth). Models were constructed of lucite, which has acoustic velocities consistent with analytically estimated velocities of sea ice ridges, and of "fast cast," a material with much lower velocities. For $ka > 1$ measurements from randomly oriented and parallel lucite and fast cast ridges on a thin plate and on an infinite half-space were nearly identical, and in good agreement with Twersky theory prediction. At the lowest frequency, $ka = 0.67$, and at grazing angles less than 20° , measured reflection losses were significantly greater than predictions: twice as large for randomly oriented lucite ridges on a thin plate. The phase change upon reflection was strongly dependent on the material properties of the ridges and their orientation. Results for randomly oriented lucite ridges were found to be reasonably consistent with previously reported inferences of reflection loss and phase based on matched field processing of data recorded on a large aperture vertical array in the Arctic [Livingston and Diachok, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S85 (1987)].

2:18

M13. Low-frequency transmission loss modeling in the Arctic using measured mode attenuation coefficients. T. C. Yang (Naval Research Laboratory, Washington, DC 20375)

Low-frequency transmission loss in the Arctic can be effectively described by the reflection/scattering loss at the ice-water interface due to the upward refractive sound-speed profile; the sound field that interacts with the bottom is quickly attenuated as to be unimportant at long ranges. Various models for the reflection/scattering loss have been proposed in the literature and can be shown to agree with single hydrophone data by adjusting the ice roughness or other parameters in the model. However, a critical test of the model predictions has to be conducted using vertical array data, since the depth dependence of the sound field provides a more meaningful measurement of the angular dependence of the reflection/scattering loss. In this paper, the angular and frequency dependence of the reflection/scattering coefficients was investigated, based on the mode attenuation coefficients previously deduced from vertical array data. It was found that the measured reflection/scattering loss was larger than most model predictions using realistic ice roughness parameters. A transmission loss model, based on a simple parametric fit to the reflection/scattering data, was shown to agree well with previously published data.

M14. Sea ice reflection coefficient estimates at low frequency. George R. Giellis and T. C. Yang (Naval Research Laboratory, Washington, DC 20375-5000)

Results of a sea ice reflectivity experiment conducted at the FRAM IV ice camp will be presented. A vertical array with 27 receivers at depths ranging from 30–960 m was employed. Impulsive sources at various depths (30–120 m) were set off at a range of 1145 m from the receivers. Analysis of direct and reflected pulses at the receivers resulted in reflection coefficient estimates at frequencies in the 50 Hz to 1 kHz range for reflection angles between 6° and 60° . The results will be compared with previous data [T. C. Yang and C. W. Votaw, *J. Acoust. Soc. Am.* **70**, 841–851 (1981)] and model calculations.

2:26

M15. Two numerical models—one simplistic, the other sophisticated—applied to under-ice acoustic propagation and scattering. Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529-5004) and Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148)

Many factors make modeling under-ice acoustic propagation and scattering difficult: Interfaces are rough with random protrusions; ice is acoustically penetrable; little data exist; few validated models exist; frequencies of interest range from the hertz to the kilohertz region. Thus no one model appears sufficient for all conditions. With this in mind, two under-ice acoustic models at opposite extremes are examined. One model is simplistic, using Kirchhoff approximation, ice reflection coefficients [S. A. Chin-Bing, *J. Acoust. Soc. Am. Suppl.* 1 **78**, S57 (1985)] and rms roughness. Model results compare favorably with high-frequency measurements from a flat cylindrical ice block. Implications are that simplistic models may be useful in generating synthetic under-ice acoustic data to augment sparse experimental data bases. The other model is sophisticated in that it is a full-wave range-dependent acoustic propagation model based on the finite element method [J. E. Murphy and S. A. Chin-Bing, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S9 (1987)] and applied to an under-ice surface simulated by fractals superimposed on an ice keel representation. Numerical results are presented for this low-frequency model. [Work supported by ONR/NORDA.]

2:30

M16. Low-frequency propagation across the East Greenland and Frontal Zone: Depth dependence of transmission loss. Leonard E. Mellberg, D. N. Connors, D. G. Browning, G. Botseas (Naval Underwater Systems Center, Newport, RI 02841), and O. M. Johannessen (Geophysical Institute, University of Bergen, Bergen, Norway N-5014)

In a previous paper [L. E. Mellberg *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **78**, S71 (1985)] the directional dependence of acoustic modes across the East Greenland Frontal Zone was examined. However, for a given source and receiver depth, transmission loss can be due to a combination of these directionally dependent acoustic modes. By changing the depths of the source and receiver, the directional dependence of the transmission loss may be changed, in some cases to the extent of causing a reversal in the dependency. Results are given for a series of source and receiver depths located at various ranges relative to the frontal zone. [Work supported by NUSC and ONR.]

2:34

M17. Relative scattering from dry cracks, ice ridges, and leads. Jacques R. Chamuel (Sonoquest/Advanced Ultrasonics Research, P.O. Box 153, Wellesley Hills, MA 02181) and Gary H. Brooke (Defence Research Establishment Pacific, FMO, Victoria, British Columbia VOS 1B0, Canada)

Little is known about the characteristics of dry cracks in Arctic sea ice and their effect on low-frequency underwater acoustic waves interacting with the ice. Ultrasonic modeling results are presented showing the relative scattering from dry cracks, leads, and ridges in floating plates bounding a long shallow-water waveguide. Backscattering from solid ridges and ridges with internal cracks reveal the potential importance of dry cracks on low-frequency transmission loss prediction. In models with a large

number of rectangular ridges, great transmission losses were observed at frequencies nearly coinciding with twice the ridge width. The article focuses on multiple scattering from infinitely long shallow parallel cracks, where the crack depth is much smaller than the wavelength. For a given number of cracks or ridges, the attenuation of low frequencies propagating in the waveguide is greatly affected by their spatial distribution along the waveguide boundary. [Work supported by DREP and ONR.]

TUESDAY AFTERNOON, 17 NOVEMBER 1987

POINCIANA ROOM, 2:00 TO 4:15 P.M.

Session N. Architectural Acoustics II: Acoustical Design of Renovated Performance Facilities II

David Lubman, Chairman

David Lubman and Associates, 14301 Middletown Lane, Westminster, California 92683

Chairman's Introduction—2:00

Invited Papers

2:05

N1. The impact of Playhouse Square on the resurgence of Cleveland. Lawrence Wilker (Playhouse Square Foundation, 1501 Euclid Avenue, Suite 810, Cleveland, OH 44115)

Cleveland, Ohio, was a city in economic distress when a group of farsighted citizens and local officials decided to develop a downtown cultural center. Lacking the funds for a 300 million dollar project, it was decided to renovate a three-room vaudeville/film theatre complex, known as Playhouse Square. One theatre was designated as a drama/dance/chamber orchestra facility, another as the home of the Cleveland Opera and Ballet Companies, and the third as a venue for popular entertainment. The paper traces the history of the project and its impact on the resurgence of Cleveland.

2:30

N2. New theatres for old. Norris Strawbridge (Borrelli Frankel Blitstein Sasaki Associates, Inc., 7000 S.W. 62nd Avenue, Suite 520, Miami, FL 33143)

The Miami Beach Theatre for the Performing Arts, recently renamed The Jackie Gleason Theatre, began its life as a multipurpose theatre/sports facility. In the mid-1960's a combination of community needs and a decision by Jackie Gleason to broadcast his television programs from Miami Beach resulted in a renovation of the hall as a performing arts/television studio venue. Although the theatre lacked certain amenities, it fostered the growth of one of the largest concert/opera/dance series in America and has become the home for a 26-week Broadway show season. Recognizing the inherent problems in the facility, the City recently commissioned a second renovation to bring the space up to contemporary acoustic and theatre standards. The paper discusses the methods used to develop an architectural program for the renovation and the solutions developed by the design team to attain desired goals.

2:55

N3. Renovation of the Orpheum Theatre, Vancouver, Canada. Ewart A. Wetherill (Wilson, Ihrig and Associates, 5776 Broadway, Oakland, CA 94618)

Despite the ever-increasing cost of performance facilities, the design and construction of a completely new auditorium is preferred with remarkable consistency to the alternative of renovating or remodeling an existing unused, or under-used, auditorium. However, the reuse of an existing hall may yield—in addition to substantial cost savings—such advantages as prime location, existing social recognition, and historical significance. This paper reviews the remodeling of the Orpheum, originally a vaudeville theatre, from cinema to concert hall as an alternative to its conversion to several small cinemas. In most respects, the renovation has preserved the original architectural character while attaining a critically acclaimed performance facility.

N4. Acoustics of the new Filene Center, Wolf Trap Farm Park for the Performing Arts. Thomas R. Horrall and Rein Pirn (BBN Laboratories, Incorporated, 125 CambridgePark Drive, Cambridge, MA 02140)

To replace the original Filene Center that was destroyed by fire, a new structure was designed and built in a scant 2 years (1982–84). Like its predecessor, the 3800-seat covered house, with an additional 3000 seats on the lawn beyond, serves as a summer home for the National Symphony Orchestra and caters to many other forms of artistic production. Under a mandate to rebuild the structure as it had been, the acoustical options were limited. Known problems such as discrete echoes off the high ceiling were eliminated. Stage acoustics were greatly improved by incorporating a totally new shell. Most significantly, the center was equipped with state-of-the-art sound systems, including a sophisticated enhancement system that covers the under-balcony areas. The principal acoustical characteristics of the center are discussed and the sound systems, their purpose, and their operational parameters described.

3:45–4:15

Panel Discussion

TUESDAY AFTERNOON, 17 NOVEMBER 1987

TUTTLE NORTH ROOM, 2:00 TO 4:05 P.M.

Session O. Structural Acoustics and Vibration II: Fluid-Loaded Structures

Albert J. Tucker, Chairman

Office of Naval Research, Mechanics Division, Code 1132SM, 800 North Quincy Street, Arlington, Virginia 22217

Chairman's Introduction—2:00

Contributed Papers

2:05

O1. Simulated wave-vector filtering of impact-generated flexural wave disturbances on a finite thin-walled cylinder. Allan D. Pierce, Jerry H. Ginsberg, and Gary P. Schwaiger (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The authors have recently undertaken a theoretical simulation of experiments similar to those performed recently at NRL by Earl G. Williams and his colleagues, in which a submerged cylinder is transiently excited and normal displacements are inferred using the generalized near-field acoustical holography (GENAH) technique. In the present paper, the cylinder is modeled as a thin-walled shell with rigid immovable end caps and with the end conditions taken as being that of simply supported. A short duration point force $f(t)$ is applied normally to the surface and one seeks to describe the resulting normal displacement field using a high-frequency (and short wavelength) asymptotic theory of propagating and evanescent flexural wave pulses on a curved plate. Fluid loading is taken into account approximately with modifications derived using the assumption that the flexural wavelengths are short compared to the length and radius of the cylinder. One asks for the time and surface position dependence of the normal surface displacement within a local region of the cylindrical surface before and after a simulated wave vector filtering has been applied. The filtering process is such that one seeks to isolate only that contribution from a narrow frequency band and from a localized region of wavenumber space. The exercise is intended to demonstrate that one can isolate and follow traveling wavelets propagating away from the source, spiraling around the cylinder, and undergoing successive reflections at the end caps. [Work supported by ONR.]

2:20

O2. Numerical implementation of variational techniques for sound radiation from axially and radially vibrating finite cylinders. Xiao-Feng Wu (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Previous work by the author with Pierce and Ginsberg [J. Acoust. Soc. Am. Suppl. 1 79, S35 (1986)] using a variational formulation has been extended to axisymmetric bodies in a general state of axisymmetric vibration. A Rayleigh–Ritz technique is used to calculate the acoustic pressure distribution on the surface of axisymmetrically vibrating finite cylinders. The computed surface acoustic pressure and the given normal surface velocity are then taken as inputs into an integral relation for determination of the farfield radiation patterns. Different trial functions are employed in the expansion for the unknown surface pressure and results are compared with each other. It is demonstrated that the computational solutions converge much faster toward the correct answers if one incorporates his physical insight and prior knowledge into the construction of admissible trial functions for the unknown surface pressure. Examples studied include finite cylinders vibrating, respectively, in the radial direction and axial direction as a rigid body. Numerical results agree well with those previously published by Copley and by Fenlon and with those computed using Rogers' SHIP program. [Work supported by ONR.]

2:35

O3. Impulse response of a continuously ribbed panel. G. Maidanik and J. Dickey (David Taylor Naval Ship Research and Development Center, Bethesda, MD 20084)

The impulse response function of a uniform panel immersed in a uniform environment is assumed known. The modification in this impulse response function due to the attachment of identical and regularly spaced ribs is derived. The manner of negotiating this modified impulse response function from the discreteness to the continuum of ribs is discussed. The negotiation reveals the criteria that need to be satisfied for uniformly lumping the ribs in the panel. When the criteria are substantially satisfied, the ribbed panel becomes essentially a uniform panel. The ribs are then accounted for by lumped modification of the uniform impedance of the unribbed panel.

2:50

O4. Vibration of a plate with distributed structural stiffness. Mauro Pierucci (Department of Aerospace and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

In most analyses of plates, only two types of plate vibrations have received wide attention: homogeneous plates and plates with discrete discontinuities. In this work, the structural inhomogeneity has been selected to be continuous and distributed. The plate is assumed to be infinite. The stiffness has been assumed to vary in a sinusoidal manner. The resulting differential equation has been solved by the use of Fourier transforms and the solution in the wavenumber space is presented. The plate admittance is a function of the structural wavenumber and the wavenumber of the distributed inhomogeneity. The results for the *in vacuo* and the fluid-loaded plates are presented. In the very short and the very long inhomogeneity wavelength limits, the effect of the structural inhomogeneity is very weak at all structural wavenumbers. The main contribution to the structural admittance is present when the value of the structural inhomogeneity wavenumber is close to the coincidence structural wavenumber. For the *in vacuo* cases, it is shown that the effect of the inhomogeneity is limited to values of the structural wavenumber equal to the coincidence wavenumber plus or minus an integral multiple of the inhomogeneity wavenumber.

3:05

O5. Resonant response of elastically supported, fluid-loaded plates. Jamel Hammouda and Courtney B. Burroughs (Engineering Science and Mechanics Department, The Pennsylvania State University, University Park, PA 16802)

An analytic model of the response of finite plates simply supported on two opposing edges and elastically supported on the other edges is derived. Transverse shear and rotary inertia in the plate and adjustable stiffness constants in the elastic supports are included in the model. The elastic support stiffness constants are adjusted to simulate simply supported, clamped, and free boundary conditions, and the results are compared to published results. The variation of resonant frequencies with the stiffnesses of the elastic supports is shown. The effect of fluid loading on the resonant responses of the elastically supported plate is presented.

3:20

O6. Dynamic response of submerged layered composites. Arthur Braga and George Herrmann (Division of Applied Mechanics, Department of Mechanical Engineering, Stanford University, Stanford, CA 94305)

The dynamic interaction between elastic and homogeneous two-phase periodically layered bodies and a surrounding acoustic fluid has been studied. The plane-strain elastodynamic behavior of the composite has been modeled on the basis of Floquet's theory for periodic structures. This approach consists of solving the equations of motion for a cell that repeats periodically, and then applying Floquet's theorem to extend the solution

to the entire body. The dispersion spectrum exhibits the classical Brillouin structure of stopping and passing bands. Even though Floquet's theory applies essentially to infinite regions, it is nevertheless possible to adapt the scheme for finite boundaries and to formulate interface conditions with an acoustic fluid. Qualitative and quantitative results are presented for harmonic motions of two geometries, namely, a layered half-space with an overlying acoustic fluid and a submerged infinite laminated plate. Special attention is paid to the energy partition between the layered solid and the fluid for the complete range of frequencies. [Work supported by ONR.]

3:35

O7. Accelerometer array as a means to evaluate structural underwater sound. D. Brenot (Department of Physics, Onera, BP 72, 92322 Châtillon Cedex, France)

The turbulent boundary layer low-wavenumber component, an important source of underwater structure excitation, cannot be precisely determined. Nevertheless, the resulting wall accelerations can be accurately measured. They already contain the coupling with the fluid, avoid the computation of the structural motion, and allow direct calculation of the farfield noise radiated by the vibration itself: The single frequency farfield noise radiated in a given direction by a baffled panel comes from one component of the wavenumber acceleration spectrum, and only one (see M. C. Junger and M. Péruilli, *Éléments d'Acoustique Physique*, p. 59). This component may be obtained experimentally by performing the Fourier transform of the wall acceleration using an accelerometer array. An underwater experiment is shown on a 3-mm-thick panel with a 5×5 accelerometer array matrix ($\Delta x = 70$ mm, $\Delta y = 60$ mm) and on a thick cylinder (radius: 1 m; length: 12 m) with a 10-accelerator linear array. The parabolic laws between wavenumber and frequency clearly appear and are quite typical of each element. These measures can thus be used to estimate their radiation efficiency. [Work supported by DCN/CER-DAN.]

3:50

O8. New approaches in vibroacoustic measurement system design. Joseph A. Clark and Paul M. Honke (David W. Taylor Naval Ship Research and Development Center, Bethesda, MD 20816 and Catholic University, Washington, DC 20064)

Recent advances in measurement instrumentation and in computer hardware such as more fully programmable signal conditioners, faster data converters, processors and other handlers, higher resolution displays, and larger memories invite the development of new vibroacoustic measurement approaches. In this talk, the design and implementation of a new system for measuring radiated and self-noise paths associated with complex, large-scale, immersed structures will be described. New approaches will be presented in the context of software developed or utilized to achieve improved performance of control, processing, display, and analysis functions. Advantages of the new approaches for studies of more subtle and complex vibroacoustic effects will be discussed.

Session P. Speech Communication III: Recognition

Jay G. Wilpon, Chairman
AT&T Bell Laboratories, Murray, Hill, New Jersey 07974

Chairman's Introduction—8:10

Contributed Papers

8:15

P1. Performance benchmarks for isolated word recognition. Lawrence R. Rabiner and Jay G. Wilpon (AT&T Bell Laboratories, Murray Hill, NJ 07974)

The performance of isolated word speech recognition systems has steadily improved over time as more is learned about how to represent the significant events in speech and how to capture these events via training algorithms. In particular, algorithms based on both template matching and hidden Markov models (HMMs) have been developed that yield high accuracy on several standard vocabularies including the ten digits (zero to nine) and the set of letters of the English alphabet. Results are presented here that show the effect of analysis parameters (e.g., frame size, frame rate) on recognition performance for a speaker trained system and the performance of both template and HMM-based recognizers (both speaker trained and speaker independent) on the digits and the alphabet vocabularies.

8:30

P2. Classification of fluent speech in coarse acoustic classes. Paul-Eric Stern (Intelligent System Laboratory, Xerox Palo Alto Research Center, 3333 Coyote Hill Road, Palo Alto, CA 94304)

Broad phonetic classes have been demonstrated to be useful for constraining search during lexical access but necessitate that the recognition of broad phonetic features be achieved in an error-free mode. This goal is not realistic due to the abstract definition of phonetic classes. However, a similar task may be performed through the recognition of acoustic classes that are subsequently mapped to phonetic categories. A coarse acoustic recognizer has been implemented in the context of a multispeaker fluent speech recognition system. It furnishes robust acoustic representations corresponding to different degrees of granularity of speech that will constrain further processing on the signal. This classifier has been implemented using a Bayesian technique allowing the application of different kinds of class recognition decisions. The parameters used to represent the speech signal are dependent on the acoustic classes and range from whole spectrum characteristics to specific discriminating features. The choice of acoustic classes is discussed, one main criterion being the maintenance of a correspondence between acoustic and phonetic labels. Because of the variability of the signal, but also of the propensity to error of any recognizer, multiple and contextual mapping is advocated. Results are reported and commented upon. [Work supported in part by INRIA and DARPA.]

8:45

P3. Using features to empirically generate pronunciation networks. Alexander I. Rudnicky (Department of Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213)

The construction of high-quality lexical models for speech recognition systems is a labor-intensive process, requiring not only a great deal of time but also appreciable expertise on the part of the human model builder.

Techniques that automate all or parts of this process are therefore highly desirable. Two approaches have been used: the generation of pronunciation networks from baseforms by application of phonological rules and the abstraction of such networks from transcription data. This paper describes a semiautomatic procedure for deriving lexical models from manual phonetic transcriptions. The algorithm uses a feature-based distance metric to align different pronunciations and produces phonetic networks that specify alternate pronunciations for words. The paper describes the results of constructing lexical networks for a 1000-word vocabulary, using a corpus of 10 000 work tokens. It should be noted that the technique described here is not restricted to manual transcriptions, but can be extended to automatic transcription such as the one produced by a front-end component of a speech recognition system.

9:00

P4. A word classification method for the recognition of Japanese from a large vocabulary. Kazunaga Yoshida (C&C Information Technology Research Laboratories, NEC Corporation, 4-1-1, Miyazaki, Miyamaeku, Kawasaki 213, Japan)

This work is motivated by results of recent studies which show that lexical access using robust broad phoneme classes can potentially result in small sets of candidate words from a large lexicon, and thus is effective for large vocabulary word recognition. In this paper, a set of broad phoneme classes suitable for Japanese and a new multilevel lexical access model are proposed. First, broad phoneme classes that have robust features for Japanese are investigated, and their constraining power is examined using a dictionary containing over 20 000 words. Next, a multilevel lexical access model is proposed based on our study about the broad phoneme class. The model consists of four stages: syllable nucleus detection, consonant classification, syllable nucleus classification, and detailed phoneme classification. The lexical search space can be significantly reduced using the access model. Finally, a word classification method based on the access model is also proposed. Results of recognition experiments were encouraging. [This study was conducted while the author was visiting at the Research Laboratory of Electronics, MIT, Cambridge, MA 02139.]

9:15

P5. Application of hidden Markov models to automatic speech endpoint detection. J. G. Wilpon and L. R. Rabiner (AT&T Bell Laboratories, Office 2C-573, Murray Hill, NJ 07974)

Accurate detection of the boundaries of a speech utterance during a recording interval has been shown to be crucial for reliable and robust automatic speech recognition. The endpoint detection problem is fairly straightforward for high-level speech signals spoken in low-level stationary noise environments. However, these ideal conditions do not always exist. One example, where reliable word detection is difficult, is speech spoken in a mobile environment. Currently, most endpoint detection algorithms use only signal energy and duration information to perform the endpoint detection task. In this paper, an endpoint detection algorithm is

presented that is based on hidden Markov model (HMM) technology. Based on a speaker-dependent speech database from four talkers, and recorded in a mobile environment under five different driving conditions (including while traveling at 60 mph with the fan on), several endpoint detection schemes were tested. The results showed that the HMM-based approach to endpoint detection performed significantly better than the energy-based system. The overall accuracy of the system using the HMM endpoint detector, when tested on the 11 word digits vocabulary (*zero* through *nine* and *oh*) with speech recorded in various mobile environments, was 99.4%. A top recognition performance of 99.7%, on the same conditions, was achieved by using an HMM recognizer with explicit end-pointing.

9:30

P6. Some techniques for creating robust stochastic models for speech recognition. Chin-Hui Lee (Speech Research Department, AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Recently, hidden Markov models (HMM) have been applied successfully to both isolated and connected word recognition. However, when the same formulation is adopted for recognition of more confusable vocabularies such as English alphabets, the recognition performance is often less satisfactory. The main reason is that a more accurate model is required. Such a model should be more robust against small training sample sizes and should also be properly initialized so that the finer features used to discriminate confusable word pairs can be extracted. In this paper, three specific robustness issues will be investigated: choice of observation densities, model initialization, and incorporation of duration information. In a step-by-step attempt to address those issues, it was found that the same HMM formulation can still be adopted if acoustic/phonetic knowledge about the vocabulary is taken into account in the model parameter estimation and recognition phases. Testing on a 39-word English alpha-digit vocabulary, in a speaker trained mode, indicates that the recognition performance can be significantly improved and the results are comparable to the template-based DTW recognizer if model parameters are properly initialized and durational information is adequately incorporated.

9:45

P7. Probabilistic estimation of lexical stress. Ghassan J. Freij (Speech Laboratory, Cambridge University Engineering Department, Trumpington Street, Cambridge CB2 1PZ, United Kingdom)

A probabilistic technique is described for the automatic estimation of lexical stress pattern of isolated words using hidden Markov models (HMMs) with continuous asymmetric probability density functions. Stress information is potentially useful for large-vocabulary speech recognition in that it facilitates the extraction of more robust phonetic data within stressed syllables and leads to more efficient lexical access. Adopting a binary stressed-unstressed strategy, a constrained HMM network was trained using observation vectors consisting of ten acoustic measurements of intensity, pitch, and spectral balance. These measurements were extracted from a corpus of disyllabic stress-minimal word pairs, each word embedded in a phrase and repeated eight times. Further, the underlying temporal structure of the data is represented implicitly by the probability of remaining in a state in relation to the probability of exiting from it. Using this technique, correctness of classification was measured to be 89% for stressed syllables and 81% for unstressed syllables.

10:00

P8. Should ASR front-end be insensitive to fundamental frequency? (perceptual shift of formant position due to fine harmonic structure of voiced speech). Hynek Hermansky (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 03105)

Due to the fine harmonic structure of the source spectrum of voiced speech, most automatic speech analysis techniques estimate formant positions that systematically deviate from the resonant frequencies of the vo-

cal tract. This property is in conflict with linear source-filter models of speech production and an effort is made to minimize the effect of the fundamental frequency F_0 on the analysis result [e.g., Hermansky *et al.*, *Proceedings ICASSP-83* (IEEE, New York, 1983), pp. 778-780]. Our experiments show that the human auditory perception has properties similar to the above described properties of automatic analyses. The method of confusion scaling is applied in paired comparison of synthetic vowel-like stimuli with varying F_0 and the second formant frequency F_2 . The results show that the perceptual estimate of formant position goes through a cycle of positive and negative deviations as the relative F_2/F_0 frequency goes from one integral number to the next. The position is estimated the most accurately when the formant peak falls either into a harmonic peak or if it lies right between two harmonic peaks. The maximal deviation of the estimate due to harmonics of the F_0 from the 120-Hz neighborhood in the 2-kHz second formant region is about $\pm 0.5\%$. Our findings imply that when modeling the speech perception, as in the automatic speech recognition, a certain degree of sensitivity of the analysis technique to F_0 might be beneficial.

10:15

P9. The importance of context on the realization of phonemes. Francine R. Chen (Xerox PARC, 3333 Coyote Hill Road, Palo Alto, CA 94303)

Linguists and speech researchers have formulated many rules describing the variability in the phonetic realization of spoken words. Using a data base of spoken sentences that has been phonetically transcribed (Lamel *et al.*, "Speech database development: Design and analysis of the acoustic-phonetic corpus," DARPA Speech Recognition Workshop, 1986), some of these rules are quantified by examining the effect of different phonemic environments on the phonetic realization of spoken words relative to a canonical dictionary representation. Modifications are analyzed in terms of the general categories of insertion, deletion, substitution, or match, and in detail in terms of each type of modification. The different contexts are studied along many dimensions, including stress, foot position, part of syllable, and word-initial/word-final position. Representations for compactly describing the environmental contexts are also developed. Preliminary results suggest that the importance of each of the dimensions that can be used to specify a phoneme's context is phoneme-dependent. Additionally, the strength of the relationship between observed modifications and contexts vary across dimensions for each phoneme. [Work supported in part by DARPA.]

10:30

P10. An integrated approach to automatic detection of retroflexed and high-fronted approximants in continuous speech. Roy Gengel and James Hieronymus (Institute for Computer Sciences and Technology, Bldg. 225, A216, National Bureau of Standards, Gaithersburg, MD 20899)

Our goal is the development of speaker-independent, semivowel detector classifiers for use with continuous speech [R. W. Gengel and J. L. Hieronymus, *J. Acoust. Soc. Am. Suppl.* 1 **80**, S19 (1986)]. Progress on detection of retroflexed approximants ($/r/,/ɜ/,/ə/$) in one branch of a classification tree, and high-fronted approximants ($/y/,/i/$) and high-front vowels ($/i/,/e/,/ɛ/$) in another, is reported. The analytical scheme relies heavily on energy ratios selected on the basis of "frames of reference." For example, a basic frame of reference is the energy in the frequency band between approximately 2-3 kHz. According to formant theory, the energy level in this band should be relatively low for retroflexed approximants (F_1-F_3 below 2 kHz) and relatively high for high-fronted approximants (F_2, F_3 above 2 kHz) for male speakers. In contrast, the energy band between about 1.2-2 kHz should be relatively high for retroflexed approximants while the band between about 0.6-1.6 kHz should be low for high-fronted approximants. By using "energy 2000-3000" as part of the denominator for a retroflexion detector and as a part of the numerator for a high-fronting detector, energy ratios are developed. The quotients of these ratios are then used in two-dimensional cluster analyses. Other ratios are used in subsequent cluster analyses. Results so far indicate that greater than 90% of target tokens can be correctly detected in a corpus of several hundred read sentences. Ratios of correct detections/

incorrect detections vary between approximately 1.0/0.63–1.0/2.50. Some pros and cons of the method will be presented. [Work supported in part by DARPA Contract N0003984PD41304.]

11:15

10:45

P11. Across-speaker and within-speaker variability of British English sibilant spectral characteristics. P. F. D. Seitz, R. A. W. Bladon, and I. M. C. Watson (Phonetics Laboratory, University of Oxford, 41 Wellington Square, Oxford OX1 2JF, England)

Testing hypotheses regarding the potential as phoneme discriminators of certain properties of "physical" and "auditorily transformed" sibilant spectra [A. Bladon and F. Seitz, *J. Acoust. Soc. Am. Suppl.* 1 80, S19 (1986)], the present study draws on a database of 20 male and 20 female talkers, containing over 2000 tokens of /s/, /ʃ/, /z/, and /ʒ/ in a variety of phonetic contexts and speaking styles. Two automatic parameter extraction routines were used to obtain measurements, for each token, of physical and auditorily scaled values of the main spectral peak, gradients of the leading and trailing spectral edges, and bandwidth of the peak. Especially in females' productions of the apicals, large and characteristic variations in these parameters were observed as correlates of individual talker, phonetic context, and style. Overviews of the envelopes of physical and auditory variation are given; based upon this knowledge, a speaker-adaptive strategy for discrimination of British English sibilants is proposed. [Work supported by the Alvey Programme.]

11:00

P12. Automatic detection for diphthongs. Hisao M. Chang (NYNEX Corporation, 70 West Red Oak Lane, White Plains, NY 10604)

It is generally recognized that diphthongs do not have steady states and only display identities through a dynamic shift of their spectra. In Miller's auditory-perceptual theory, dynamic shifts of diphthong spectra are represented as paths on a "vowel map" [J. D. Miller, *J. Acoust. Soc. Am. Suppl.* 1 81, S16 (1987)]. From the utterances of English digits by two male and two female native speakers, an auditory-perceptual transformation system and a segmentation rule were used to transform the spectral prominences contours derived from those utterances containing diphthongs /aɪ/ and /eɪ/ in paths on the vowel map, and it was found that all paths from the same diphthong, independent on talker's sex and phonetic contexts, travel in the same route and form a "diphthong pip." After the detailed analysis of the paths, an automatic detection algorithm was developed based on angles, durations, lengths, origins, and other attributes of the paths. When applying the algorithm to digit utterances from 21 new talkers, the recognition accuracies for /aɪ/ and /eɪ/ were 75% and 90% while the insertion rates were only 1.1% and 4.5%, respectively. [Work supported by NINCDs and done at Central Institute for the Deaf, St. Louis MO.]

P13. Speaker-independent automatic vowel recognition based on overall spectral shape versus formants. Stephen A. Zahorian and Amir J. Jagharghi (Department of Electrical and Computer Engineering, Old Dominion University, Norfolk, VA 23529)

Automatic recognition experiments were performed to compare overall spectral shape versus formants as speaker-independent acoustic parameters for vowel identity. Stimuli consisted of four repetitions of 11 vowels spoken by 17 female speakers and 12 male speakers (29•11•4 = 1276 total stimuli). Formants were computed automatically by peak picking of 12th-order LP model spectra. Spectral shape was represented using three methods: (1) by a cosine basis vector expansion of the power spectrum; (2) as the output of a 16-channel, 1/3-oct filter bank; and (3) as the output of a 16-channel mel-spaced filter bank. Automatic recognition was based on maximum likelihood estimation in a multidimensional space. For all cases considered, the representations based on spectral shape resulted in significantly higher recognition accuracy than for recognition based on only three formants. For example, using the entire database of all speakers and 11 vowels, recognition based on spectral shape was about 85% vs 69% for three formants. If the data were restricted to female speakers and the seven vowels /a,i,u,æ,ɜ,i,ɛ/, recognition was about 97% based on spectral shape versus 84% for formants. These results indicate that, at least for automatic recognition of vowels, spectral peak detection is neither necessary nor sufficient. [Work supported by NSF.]

11:30

P14. Formant tracking using statistical pattern recognition. Robert T. Gayvert and James Hillenbrand (Rochester Institute of Technology Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester, NY 14623)

Formant trackers that use peak-picking algorithms and *ad hoc* rules tend to suffer from occasional severe mistakes in labeling formants. Recently, Kopec [IEEE Trans. Acoust. Speech Signal Process. ASSP-34, 709–729 (1986)] used a statistical approach based on hidden Markov models to avoid these kinds of problems. This study uses another statistical technique based on discriminant analysis. A model is trained on hand-marked formant trajectories and various features from a collection of utterances. The same features are then extracted from a test utterance and used to determine the most probable formant frequencies. Any combination of features, including peaks from LPC and DFT spectra, can be used with this technique. Preliminary results indicate that this method avoids serious mistakes. Results of evaluating the formant tracker on a large number of training and test sets will be presented. [Work supported by Rome Air Development Center under contract F3060285-C-0008.]

WEDNESDAY AM

Session Q. Engineering Acoustics II: Special Session on Directional Microphone Systems for Airborne Sound

James E. West, Chairman

AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, New Jersey 07974-2070

Chairman's Introduction—8:30

Invited Papers

8:35

Q1. A unitary multielement ultrasonic array for focused imaging in air. Leslie Kay (220 Boylston Street, Unit 9007, Boston, MA 02116)

A condenser microphone is described that incorporates 20 independent conductive elements coupled to a single laminar diaphragm. The elements are separately excited by an electrical signal, delayed in time relative to adjacent elements, to enable the formation of a focused beam within the close nearfield of the array. Using a radiation frequency covering an octave bandwidth from 100–200 kHz, a spatial resolution of 0.2 in. parallel to the aperture and 0.1 in. perpendicular to the aperture at the point of maximum focus has been achieved. No aperture shading has been incorporated at this stage so that the diffraction field can be explored. Since this is formed by a very wide bandwidth radiation, an unusual field plot was expected. The results are discussed and a video shows examples of the high resolution real time imaging in air that can now be achieved. This has many applications, particularly in automation as an alternative to optical vision when distance to an object is essential information.

9:00

Q2. A review of adaptive microphone arrays. M. M. Sondhi (Room 2D-536, AT&T Bell Laboratories, Murray Hill, NJ 07974)

In this talk, several algorithms for adaptive optimization of broadband microphone arrays will be described. The arrays may be linear, two-dimensional, or three-dimensional. Each microphone in the array feeds a tapped delay line filter, and the outputs of the filters are summed to give the final output. The problem to be discussed is that of adaptively adjusting the tap weights of the filters in order to optimally receive a desired speech signal in the presence of broadband noise sources with unknown locations and possibly slowly varying characteristics.

9:25

Q3. Directional processing for interference reduction in hearing aids. W. M. Rabinowitz and P. M. Peterson (Research Laboratory of Electronics, MIT, Room 36-761, Cambridge, MA 02139)

Listeners with hearing impairments often have great difficulty in interpreting acoustic environments having multiple sources and/or reverberation. Directional processing, obtained by weighting the outputs of multiple sensors, offers potential for reducing such interference, although cosmetic considerations impose fundamental limitations on performance. In *fixed* processing, weights are held constant to achieve a fixed antenna pattern (pointing straight ahead) and are chosen as a compromise between classical and supergain limits so as to maximize directivity with tolerable sensitivity to receiver noise. A particular second-order gradient microphone system, with a sensor span of 4.1 cm and formed simply by subtracting the outputs of two cardioid capsule hearing-aid microphones, exhibits a directivity of 8.5 dB, which is near optimum (9.0 dB) for its noise sensitivity. In *adaptive* processing, weights are continually adjusted according to properties of the environment. An adaptive algorithm using two omnidirectional microphones separated by 20 cm (\approx headwidth) has been evaluated with a target talker straight ahead and an interference source at 45° in three computer-simulated environments with zero, moderate, and severe reverberation. Both analytic measures and intelligibility tests indicate substantial reductions in interference in zero and moderate reverberation and no reduction, or degradation, in severe reverberation. Additional and more realistic evaluations of these fixed and adaptive systems are required to determine their suitability for practical hearing aids. [Work supported by NIH.]

9:50

Q4. Automatic speech-seeking microphone arrays for teleconferencing. G. W. Elko (Room 2C-574, AT&T Bell Laboratories, Murray Hill, NJ 07974)

The effects of reverberation on speech transduction in large rooms can severely limit the intelligibility of the desired speech signal. In isotropic acoustic fields, aside from locating the microphone in close proximity to the

speech source, it is necessary to develop highly directional microphone systems. Concomitant with microphone array designs for teleconferencing is the need to steer the array at the desired speech source in order to maximize the direct-to-reverberant signal energy. The problem that will be discussed is that of designing broadband (speech bandwidth) microphone arrays that are capable of detecting a speech source and automatically steering the array to the direction of the source.

Contributed Papers

10:15

Q5. Three-dimensional microphone arrays. J. L. Flanagan (Information Principles Research Laboratory, AT&T Bell Laboratories, Murray Hill, NJ 07974-2070)

Low-cost, high-quality microphones and integrated electronics prompt examination of sophisticated arrays for sound pick-up in large enclosures. A three-dimensional, uniform configuration of receivers is studied and shown to have several benefits. Beamforming and delay steering can be accomplished over the 4π solid angle without spatial ambiguity. Beamwidth is essentially independent of steering direction. As in arrays of fewer dimensions, usable bandwidth is governed at the upper frequency limit by receiver spacing and at the lower frequency limit by linear dimensions. Harmonic nesting in three dimensions can extend the bandwidth of unique spatial discrimination to any desired range. Range selectivity, additional to unique directional discrimination, can be realized by delay steering to nonplanar wave fronts.

10:35

Q6. The evaluation of source strength using a microphone array over a hard reflecting surface. Stewart A. L. Glegg and Richard C. Smart (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

This study has been undertaken to measure the noise source height on highway vehicles. A vertical seven-element microphone array is used and the method of analysis is based on the source breakdown technique developed by Fisher and Tester. This is capable of giving narrow band or $\frac{1}{3}$ -oct source spectra for each individual source in a distribution, with resolution down to zero frequency, providing the approximate source positions are known, and there is a high signal-to-noise ratio. This method has been evaluated both in an anechoic chamber and in an outdoor environment over a hard surface. Good results have been obtained for sources which are 0.13 m apart down to 125 Hz. However, if the source positions are not exactly specified, the lower limiting frequency is increased to 500 Hz. [Work is sponsored by the Florida Department of Transportation.]

10:55

Q7. Characterization of the envelope protecting an acoustic detection system. D. Vaucher De La Croix, M. Lathuilière, and L. Périé (Metravib R.D.S., 64 Chemin des Mouilles, 69130 Ecully, France)

In a previous paper, a scheme for computing the radiation perturbation due to a two-dimensional envelope of arbitrary shape enclosing an acoustical source was presented [D. Fichot and J. M. Parot, D.3-5, 12th ICA, Toronto (1986)]. The formulation associates structural finite elements and acoustic integral equations. Though originally aimed to solve fluid-structure interaction problems with strong coupling between transducer and envelope for ka values up to 30 (k = acoustic wavenumber, a = half-largest dimension of the envelope), the code named ABER2D was applied to determine the influence of the envelope shape and/or material on the directivity of the enclosed transducer with air as ambient fluid. Deviations from the free field reference configuration have been computed at low ka values (typically $ka \approx 1$), starting from a thin circular envelope and progressively flattening it to elliptical shape. Such calculations have oriented choices concerning geometry and certain material characteristics and resulted in designing a flattened polypropylene protection envelope adapted to an acoustic detection system, which displays good transparency in the frequency band of interest. [Work supported by SEFT.]

11:15

Q8. Unidirectional, second-order gradient microphone. J. E. West, G. M. Sessler,^{a)} and R. A. Kubli (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A second-order gradient microphone with unidirectional characteristics is described. The microphone consists of two commercially available, first-order gradient electret microphones that are properly baffled and arranged with their rotational axes in line. The output of one of these transducers is subtracted from the delayed output of the other transducer. The unidirectional microphone shows a directional characteristic that is relatively frequency independent, has a 3-dB beamwidth of the mainlobe of ± 40 degrees, and exhibits sidelobes 10–20 dB below the mainlobe. After equalization, the frequency response of the microphone in its direction of maximum sensitivity is within ± 3 dB between 0.3 and 4 kHz. The sensitivity of the microphone, including its equalization, is -60 dBV/Pa at 1 kHz and the equivalent noise level for the above frequency range amounts to 28 dB linear or 20 dB(A).^{a)} Permanent address: Technical University of Darmstadt, 6100 Darmstadt, West Germany.

Session R. Psychological and Physiological Acoustics III: Temporal Perception and Comodulation

Robert H. Gilkey, Chairman

*Research Department, Central Institute for the Deaf, 818 S. Euclid, Saint Louis, Missouri 63110***Contributed Papers**

8:30

R1. Discrimination of time-expanded tonal sequences. R. D. Sorkin and M. P. Snow (Department of Psychological Sciences, Purdue University, West Lafayette, IN 47906)

Subjects were presented with two sequences of tones, each consisting of eight 50-ms sinusoidal tones. Subjects had to discriminate whether the sequences were the same or different based solely on the frequency pattern defined by each sequence of tones. The correlation between the temporal envelopes of the two sequences, the variability of sequence timing, and the length of the intersequence interval were manipulated. In the "correlated" condition, the gaps between the tones in the sequences were randomly varied across trials (sequences within a trial had identical temporal envelopes); in the "uncorrelated" condition, the envelopes varied both across and within trials. The results replicated an earlier experiment [R. D. Sorkin, *J. Acoust. Soc. Am.* (in press)]. Performance in the correlated condition was independent of temporal variability but decreased with increases in temporal variability. This pattern of results is consistent with the assumed dominance of "trace" and "context" mode processing in the two conditions. In an additional manipulation, all tone and gap durations in the second sequence of each trial were expanded by 40%. Performance in the time-expanded condition resembled that found in the correlated condition. This result is interpreted as evidence for a duration scaling process in auditory memory. [Work supported by AFOSR.]

8:45

R2. Proportional target-tone duration as a factor in the discriminability of tonal patterns. Charles S. Watson and Gary R. Kidd (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

A series of experiments on listeners' abilities to extract information from patterns with varying total durations and numbers of tonal components has previously been reported [*J. Acoust. Soc. Am. Suppl.* 1 73, S44 (1983); 77, S1 (1985)]. In those experiments listeners were tested in high-stimulus-uncertainty, same-different pattern discrimination tasks, in which the tonal patterns to be discriminated differed by changes in the frequency of one or more components. Discrimination performance in those tasks was consistent with previous measures of the frequency resolving power of the auditory system when the patterns contained one to three equal-duration components, for total pattern durations from 62.5–2000 ms. As the number of components was raised, discrimination thresholds increased by large amounts, often by factors of 10–100 for patterns with more than seven–eight components. While this result might imply an informational limit on pattern processing, it is also consistent with the hypothesis that *target tones are equally well resolved if they occupy equal proportional durations of the patterns in which they occur*. Results of a new experiment, in which proportional durations of target tones and the number of tones per pattern were independently varied, suggest that proportional duration of the target tones is in fact the primary determinant of pattern discriminability for tonal patterns ranging from 100–1500 ms in total duration. [Work supported by AFOSR and NIH/NINCDS.]

9:00

R3. Temporal aspects of the auditory system: Gap detection and the modulation transfer function. T. G. Forest and David M. Green (Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32611)

A three-stage model consisting of a bandpass filter followed by a non-linear element and a low-pass filter is often used to model temporal processing in the auditory system [e.g., N. F. Viemeister, *J. Acoust. Soc. Am.* 66, 1364–1380 (1979)]. Two experimental paradigms, the detection of gaps in noise and of sinusoidal modulation of noise, have been used to estimate the parameters of this model. A simulation of this model was used to "listen" to waveforms in gap detection and modulation detection experiments. Responses of the simulator were based on statistical calculations at the output of the low-pass filter stage of the model. Our decision statistic was the maximum-to-minimum ratio of the waveform from the low-pass filter. Good agreement between the data produced by normal-hearing human observers and the simulator occurred when the bandwidth of the initial filter was 4000 Hz and the time constant of the low-pass stage was 3 ms. [Research supported by NSF.]

9:15

R4. Illusory continuity of periodic sounds: A comparison of tonal and infratonal temporal induction. Richard M. Warren, John M. Wrightson, and Julian Poretz (Department of Psychology, University of Wisconsin—Milwaukee, P.O. Box 413, Milwaukee, WI 53201)

Tones may appear to remain on continuously when alternated with sounds having appropriate spectral composition and intensity. The limiting conditions for this illusory continuity have been used to define the characteristics of neural spectral (place) analysis of tones. The present study extends investigation of the apparent continuity of interrupted complex periodic sounds to infratonal frequencies, exploring the range of repetition frequencies from 10–2000 Hz. Evidence is presented indicating that reconstruction of obliterated segments of periodic sounds need not be based upon the neural place mechanisms associated with perception of tones. It appears that illusory continuity of periodicity can involve a perceptual synthesis based upon neural temporal information, and that explanations for the limits of illusory continuity of tones ("pulsation thresholds") solely in terms of a frequency domain analysis involving neural place information may be incomplete. [Work supported by AFOSR.]

9:30

R5. Critical bandwidth for modulation detection. William A. Yost (Parmly Hearing Institute, Loyola University, 6525 N. Sheridan, Chicago, IL 60201)

Two sinusoidal carriers separated by frequency F were each sinusoidally amplitude modulated (AM). Subjects were asked to discriminate between a complex in which the two AM signals had the same phase and a complex in which the phase spectra of the lower frequency AM components were delayed relative to the upper frequency AM components. The

threshold value of the phase shift was measured from three-point psychometric functions obtained from a two-alternative, forced-choice task. A broadband-noise background and random variation in overall level were used to force the two normal-hearing listeners to attend to the temporal structure of the two-tone AM complexes. The frequency separation between the two carriers (F), the rates of AM modulation, and the form of the phase shift (entire waveform or envelope delay) were parameters in the study. The results indicate that the threshold for detecting the phase shift is small when the two carriers are within a critical band. However, listeners are still quite sensitive to the phase shifts for very large frequency separations between the carriers. The results will be discussed in terms of the auditory system's ability to use temporal modulation to identify common sources in a complex sound. [Work supported by AFOSR-Life Sciences and the NINCDS.]

9:45

R6. Discrimination of envelope phase disparity. Gregory H. Wakefield and Brent Edwards (Department of Electrical Engineering and Computer Science, University of Michigan, Ann Arbor, MI 48109-1109)

The sensitivity of the auditory system to changes in ongoing temporal structures across different spectral regions is investigated using sinusoidally amplitude-modulated (SAM) carriers. Previous results are extended for the detection of phase disparity between the envelopes of two carriers [J. Acoust. Soc. Am. Suppl. 1 81, S34 (1987)] to the discrimination of such disparity when a nonzero relative phase offset between the envelopes is present. For a standard SAM two-carrier complex with a relative phase offset of ϕ , the relative phase disparity of the comparison $\phi + \Delta\phi$ was adjusted to measure discrimination threshold as a function of ϕ . The results show that $\Delta\phi$ varied by more than a factor of 2 as ϕ ranges from 0–180 deg. Modulation frequency (8–64 Hz) or carrier separation (500 Hz, 1–4 kHz) has little effect on this phase dependence. When expressed as a change in envelope correlation (ρ), however, we find that $\Delta\rho$ is constant as a function of ρ . Additional experiments using harmonic complexes for envelopes support the hypothesis that envelope correlation is the primary determinant of performance. A neural-based coincidence-detector model of envelope correlation is proposed to account for these findings. [Research supported by AFOSR.]

10:00

R7. Evidence for cross-channel processing in detection of envelope phase disparity. Elizabeth A. Strickland, Neal F. Viemeister, Deborah A. Fantini (Department of Psychology, University of Minnesota, Minneapolis, MN 55455), and Margery A. Garrison (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455)

The sensitivity to the correlation between the envelopes of two sinusoidally amplitude-modulated sinusoids was investigated in two experiments. Correlation was manipulated by varying the phase disparity between the envelopes. In each experiment, two carriers (either 1 and 2 kHz or 1 and 3.2 kHz) were modulated at the same frequency (8 or 128 Hz) and added together either in phase or, on signal intervals, with an adaptively determined phase disparity. The starting phase of the envelope of the lower frequency carrier was randomized across trials. In the first experiment, the lower frequency carrier was presented at 65 dB SPL and the higher frequency carrier was presented at levels from 65 to 25 dB SPL. No decrement in threshold was noted until the higher frequency signal was 20–40 dB below the lower frequency signal. In the second experiment, the lower frequency carrier was presented at 65 dB SPL to one ear, and the higher frequency carrier was presented at the same level to the other ear. Thresholds were slightly above those obtained monotonically. These results suggest that subjects are able to use information across frequency channels to detect envelope phase disparity and do not need to use envelope interactions occurring within a single channel in performing this task. [Work supported by NINCDS Grants NS12125 and NS07889.]

10:15

R8. The effect of signal-frequency uncertainty in a CMR paradigm. John H. Grose and Joseph W. Hall, III (Division Otolaryngology Head and Neck Surgery, School of Medicine, University of North Carolina, Chapel Hill, NC 27514)

The phenomenon of comodulation masking release (CMR) indicates that the auditory system is capable of comparing the outputs of several auditory channels simultaneously. The presence of the signal is presumed to be cued by a dissimilarity in the temporal pattern in that channel, relative to the remaining comodulating channels, due to the addition of the signal. In its simplest form, such a cuing mechanism could operate quite independently of any *a priori* knowledge of which channel the signal occurs in. To test this possibility, a study was undertaken to compare the effect of signal-frequency uncertainty in a comodulated and a noncomodulated background. It was hypothesized that the detection of a signal whose frequency fluctuated randomly from trial to trial would be less deleteriously affected in a comodulating background than in a noncomodulating background. A novel feature of this study was the use of amplitude-modulated pure-tone components as the background masker. Results to date indicate that the effect of signal-frequency uncertainty is itself small relative to fixed-frequency signals, and no significant difference has emerged between its measurement in comodulated versus uncomodulated backgrounds.

10:30

R9. Comodulation masking release for envelope decorrelation cue versus envelope level difference cue. Joseph W. Hall, III and John H. Grose (Division of Otolaryngology Head and Neck Surgery, School of Medicine, University of North Carolina, Chapel Hill, NC 27514)

In past investigations of comodulation masking release (CMR) it was difficult to identify the detection cue used by the listener, as several different cues were available. The present study attempted to isolate two cues: envelope correlation and envelope level. The signal was either a 10-Hz-wide noise band centered on 500 Hz, or a 500-Hz pure tone. The maskers were 10-Hz-wide noise bands centered on 400 and 500 Hz (either comodulated or noncomodulated). In one experiment the same 10-Hz-wide noise band centered on 500 Hz was used as both the signal and the masker. Here, the signal resulted in no decorrelation of envelope, but did result in an envelope level difference. In the experiment using the 500-Hz pure tone as the signal, the signal resulted in a decorrelation cue; however, any envelope level cue was rendered useless in this experiment by randomly roving the level of the flanking band between each interval, and by adjusting the noise level of the on-signal band in the dummy intervals to the level of the noise-plus-signal in the signal interval. The major result was that substantial CMRs occurred both when the envelope level cue was available but the decorrelation cue was not, and when the decorrelation cue was available but the envelope level cue was not.

10:45

R10. Comodulation detection differences with multiple signal bands. Beverly A. Wright and Dennis McFadden (Department of Psychology, University of Texas, Austin, TX 78712)

Detection thresholds were determined for signals consisting of one, two, or five noise bands embedded in eight "cue" bands. The center frequencies of the signals ranged from 1250–3250 Hz in 500-Hz steps, and those of the cue bands ranged from 500–4000 Hz in 500-Hz steps. The temporal envelopes of the cue bands were either all correlated or all uncorrelated. The envelopes of the multiple signal bands (1) were correlated among themselves as well as with the cue bands, (2) were themselves correlated, but not with the cue bands, or (3) were themselves uncorrelated. As in previous comodulation experiments, thresholds were similarly poor when the signal and cue bands were either all correlated or all uncorrelated. Detectability was typically 7–10 dB better when the signal bands were uncorrelated with the all-correlated cue bands. These results were no different for single- and multiband signals. Thus the auditory system ap-

pears better able to detect both simple and complex signals in background having the same—rather than different—temporal envelopes, as long as the temporal structure of the signal(s) is different from that of the surrounding sounds. [Work supported by NINCDs Grand NS15895.]

11:00

R11. Rate and number as limiting factors in the perception of sequences of syllables. Charles S. Watson and Blas Espinoza-Varas (Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Listeners' abilities to process the details of sequences of sounds have been shown to be limited by two major factors: the number of components in the sequence and the rate at which the components are presented. Individual differences in the ability to process large numbers of components at high presentation rates have been proposed to account for certain language- or learning-related disabilities. Little attention has been paid to the possible interactions between these two factors, nor to the distribution of these abilities in the population of normal-hearing listeners. A test was developed to measure the effects of the number of syllables and, independently, those of syllable rate, or listener's abilities to process syllable sequences. The sequences consist of two, four, or seven natural tokens of the spoken letters b, c, d, g, p, t, or v, produced as CVs ending in the vowel /i/. The duration of each token was edited to 75 ms. Syllable onset-to-onset times were 75, 150, and 300 ms. The listener's task was to transcribe the letters in each sequence, in a test format in which 180 sequences were presented, varying randomly in number and rate. Results for an initial test group show some individual's performance to be more limited by rate

rather than by number, and vice versa. Implications for the interpretation of certain language-learning disorders are discussed. [Work supported by NIH/NINCDs and AFOSR.]

11:15

R12. Informational limits in speech processing by normal and impaired listeners. Blas Espinoza-Varas, Charles S. Watson, and Mary S. Kyle (Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

It has been reported that some impaired listeners perceive speech as being "jumbled, or as going too fast," which suggests that their informational capacity may be reduced. Recently, work with nonspeech stimuli [Watson and Foyle, *J. Acoust. Soc. Am.* **78**, 375-380 (1985)] has shown that the amount of spectral-temporal information that can be extracted from sequential patterns can be limited very strongly by the number of freely varying components included in a pattern, and only weakly by the duration of the individual components. Using a variation of the speech test described in the previous paper, information capacity of normal and impaired listeners was compared for sequences of two, four, or seven syllables, presented at rates of 6.5 or 3.9 syllables/s. Percent information transmission (IT%) was estimated for each temporal position of the sequences, number of syllables, and syllable rate. The sensorineural impaired listeners had nearly uniform 45-to-55-dB losses, and ages similar to the normal listeners. Stimuli were presented at approximately constant sensation levels (30-40 dB) to minimize effects of differences in sensitivity. The maximum number of syllables yielding IT% \geq 50% was, on the average, 6-7 for normal listeners but 4-5 for the impaired listeners. [Work supported by NIH/NINCDs and AFOSR.]

WEDNESDAY MORNING, 18 NOVEMBER 1987

UMS 4 AND 5, 8:30 TO 11:45 A.M.

Session S. Underwater Acoustics III: Three-Dimensional Oceanography and Acoustical Modeling

Robert P. Porter, Chairman

Department of Electrical Engineering, FT-10, University of Washington, Seattle, Washington 98125

Chairman's Introduction—8:30

Invited Papers

8:35

S1. Three-dimensional oceanographic acoustic modeling of complex environments. W. A. Kuperman, Michael B. Porter, and John S. Perkins (Naval Research Laboratory, Washington, DC 20375)

Remote sensing combined with oceanographic modeling opens up new possibilities for quantitatively describing very large area complex oceanographic-acoustic phenomena. A complex ocean environment consists of variable oceanography including mesoscale phenomena and variable topography such as continental slopes and seamounts. Predicting the acoustical effects from any to all of these phenomena is a computationally intense problem whose efficient and accurate solution will, no doubt, ultimately employ new and innovative numerical physics. As a stimulus to developing new approaches, three-dimensional modeling results for such complex environments are being looked at using a technique that can rapidly calculate propagation loss over a wide geographical area. This approximate technique, which is based on normal mode theory, has been applied to environments that contain both variable oceanography and topography. Of prime interest in these initial studies is a search for graphical patterns that emerge for complex environments where the acoustical results reflect or "image" the oceanography and topography. It is believed that these three-dimensional representations using color graphics to display and convey complex ocean-acoustic phenomena will provide the impetus to further understanding and computing the interaction of these phenomena.

9:00

S2. Dynamical forecasting of mesoscale fronts and eddies for acoustical applications. Allan R. Robinson (Harvard University, Division of Applied Sciences, Cambridge, MA 02138)

Real-time nowcasts and forecasts of oceanic mesoscale fields (velocity, pressure, temperature, sound speed, etc.) are now feasible and have been initiated in selected regions. Predicting the "internal weather of the sea" is analogous to meteorological weather forecasting. Our approach is based on recent progress in phenomenological knowledge and modeling capabilities and is systematic [A. R. Robinson, "Predicting open ocean currents, fronts, and eddies," in *Three-Dimensional Ocean Models of Marine and Estuarine Dynamics*, edited by Nihoul and Jamart (Elsevier, Amsterdam, 1987)]; i.e., it combines dynamical model [A. R. Robinson and L. J. Walstad, "The Harvard open ocean model," *J. Appl. Numer. Math* **3** (1987)] and observational estimates via data assimilation methods. The prediction problem will be overviewed, results from an ongoing forecast scheme for the Gulf Stream (gulfcasting) will be presented, the coupling of dynamical and acoustical models will be discussed, and preliminary results from a coupled system will be presented.

9:25

S3. High-resolution ocean acoustic tomography. Peter F. Worcester, Bruce Cornuelle, Bruce Howe, and Walter Munk (Scripps Institution of Oceanography, University of California at San Diego, La Jolla, CA 92093)

The ocean acoustic tomography experiments conducted to date can be divided into *mapping* experiments, in which many sources and/or receivers are used to generate sufficient crossing ray paths to resolve the mesoscale ocean structure, and *averaging* experiments, in which data are obtained in a vertical slice between a single pair of sources and/or receivers in order to construct horizontally averaged vertical profiles. While it is economically feasible to deploy enough instruments to construct an array capable of mapping an area a few hundred kilometers on a side, it becomes prohibitive for areas one megameter and larger in scale, even though many fewer tomography instruments are required than conventional point measurements. One possible approach to make high-resolution, three-dimensional maps of large areas of the ocean is to develop a *hybrid* system, in which a combination of moored instruments and precisely positioned sources and/or receivers suspended from ships are used to generate a dense set of crossing ray paths. Simulations indicate that four or five transceivers moored in a large array to make long-term measurements of the average ocean structure could be combined with periodic cruises with ship-suspended sources and/or receivers to give excellent maps of a square megameter of ocean. It will be essential in such experiments to use data assimilation techniques to combine the data obtained as a function of time by the ship into a numerical model of the ocean. Other data types (e.g., satellite altimeter data) can be assimilated at the same time to improve the resulting maps of the ocean.

9:50

S4. The acoustic stochastic inverse problem. Terry E. Ewart (Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, WA 98105)

It has been decided that the statistics of the phase, complex amplitude, and intensity fluctuations that result from waves propagating in media with random index of refraction inhomogeneities can be predicted. In fact, the emphasis has been refocused from the development of the theory to a requirement to understand more fully the statistics of the ocean medium. An exciting result of the success of the theoretical predictions is the possibility of inverting acoustic measurements to obtain the statistics of the ocean media. In the stochastic inverse problem, one obtains coefficients of statistical models of the acoustic index or refraction using optimization techniques. These techniques require that deterministic signals, e.g., tides, be treated formally in a different manner from stochastic signals, e.g., internal waves. Examples of the inverse problem will be given from ocean and numerical experiments. Models of the oceanic internal wave, finestructure, and tidal processes, obtained from measured acoustic fluctuations, are compared with those measured. A discussion of the future expectations including 3-D considerations will be presented.

Contributed Papers

10:15

S5. Simulated reciprocal transmissions through a three-dimensional sound-speed and current structure. James A. Mercer (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98105)

A computer code that integrates Hamilton's equations for a continu-

ously defined medium was used to model a reciprocal transmission experiment along a 2000-km path. The description of the medium included the three-dimensional sound-speed and current structures of four mesoscale eddies and a western boundary jet. Eigenrays for the reciprocal transmissions are determined, displacements are compared, and the critical physical processes are identified. Travel time results are compared with those from less realistic two-dimensional and one-dimensional codes. [Work sponsored by ONT.]

10:30

S6. A discussion of three-dimensional wave propagation effects.

Ding Lee, Paul D. Scully-Power, George Botseas (Naval Underwater Systems Center, New London, CT 06320), and William L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180)

In most environments, the prediction of acoustic wave propagation is carried out with two-dimensional propagation models. This should *NOT* mislead anyone to believe that three-dimensional models are not needed. In this paper, sound-speed information from eddy models is used to examine three-dimensional effects. This examination was carried out employing the IFD (two-dimensional) and the FOR3D (three-dimensional) models. A set of numerical results from these two models will be discussed. Interesting findings will be reported. These findings provide stimulation for continued three-dimensional model development.

10:45

S7. Coherence loss modeling using the PE method. F. D. Tappert (University of Miami, Applied Marine Physics, Miami, FL 33149)

Scattering by volume fluctuations of sound speed and by boundary roughness causes a loss of coherence of propagating cw signals in the ocean. A 3-D parabolic equation model for the coherent Green's function (p) is derived that describes angle- and depth-dependent scattering loss. By integrating over a narrow sector of azimuthal angles, it is shown that a 2-D vertical plane model gives an excellent approximation for forward scattering loss. The resulting 2-D cw model is efficiently solved numerically by generalizing the split-step Fourier algorithm to perform multiple local Fourier transforms by windowing in depth, transforming from depth to vertical angle, applying an angle-dependent filter, summing over layers, and finally inverse Fourier transforming. For rough surface scattering, this method gives a loss that is linear in the grazing angle for small angles, in agreement with Marsh-Mellon theory.

11:00

S8. The PE method and the ocean tomography. E. C. Shang (NOAA, Wave Propagation Laboratory, Boulder, CO 80303)

It was well known that both the ray method and the PE method can handle wave propagation problems in 3-D environments, but both of them are approximations of the wave equation. In ray method, the diffraction term was discarded so it is good for high frequency; in PE method, the term of second-order derivation on range was discarded so it is good for narrow angle. The accumulated phase error restricts the range within which each method is valid. One has to choose a better method to adapt a

specific situation. In part (1) of this paper, the case of individual mode propagation in SOFAR channel is investigated by analyzing the different valid ranges. In part (2), the effects of a mesoscale ocean eddy on mode propagation by using PE (IFD) code are discussed. The main goal is to find out if the mode coupling matrix is sensitive enough to apply in the "matched-field" method for the purpose of ocean tomography [Work supported by NRC.]

11:15

S9. Error estimates for 3-D computational schemes of wave propagation.

George H. Knightly and Donald F. St. Mary (Center for Applied Mathematics and Mathematical Computation, Department of Mathematics, University of Massachusetts, Amherst, MA 01003)

For several computational schemes in underwater wave propagation, stability and consistency bounds are known that imply corresponding rates of convergence. It is shown, in some cases, how this information can be improved to yield explicit bounds for the truncation error between the exact solution of a continuous problem and the approximating solution generated by the computational scheme. In particular, those bounds are demonstrated for an implicit finite difference scheme in three dimensions.

11:30

S10. Three-dimensional underwater acoustic propagation through a steady shear flow.

J. S. Robertson (Department of Mathematics, U.S.M.A., West Point, NY 10996-1786), M. J. Jacobson, and W. L. Siegmann (Department of Mathematical Sciences, Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

Previous studies by the authors have incorporated current effects into parabolic approximations, when the current vector lay within the vertical source-receiver plane. Here, the current and the source-receiver plane are not constrained to be parallel, and thus a solution to a fully three-dimensional problem is obtained. A class of sound-speed profiles is examined that is advected in a steady, depth-dependent ocean current. A cw source is submerged between a horizontal pressure-release surface and a horizontal bottom. When certain asymptotic order conditions on derivatives of the current and sound speed are satisfied, parabolic approximations of canonical form result. Furthermore, if additional restrictions on azimuthal derivatives apply, the azimuthal modes are decoupled so that the azimuthal angle appears only as a parameter in the parabolic equations. Thus the pressure field may be calculated in decoupled vertical planes, i.e., in an $N \times 2$ -D approximation. Several examples are discussed, and the acoustical effects of current structure are illustrated, with emphasis on azimuthal dependence of relative intensity. [Work supported by ONR.]

Session T. Architectural Acoustics III: Analysis; Educational Facilities

Ewart A. Wetherill, Cochairman

Wilson Ihrig & Associates, Inc., 5776 Broadway, Oakland, California 94618

Richard M. Guernsey, Cochairman

Cedar Knolls Acoustical Laboratory, 9 Saddle Road, Cedar Knolls, New Jersey 07927

Chairman's Introduction—9:00

Invited Paper

9:05

T1. Calculation of free-field deviation in an anechoic room. Ji-qing Wang (Institute of Acoustics, Tongji University, Shanghai, People's Republic of China 200092) and Cai Biao (Jiaotong University of Eastern China, Nanchang, People's Republic of China)

After a review of various calculation methods currently available for free-field deviation (in dB) in an anechoic room, the authors present, in this paper, a more precise and rational calculation of the sound field by solving wave equations in consideration of the interference of first reflections only. In the case of pure tone, the phase factor plays an important role in the calculation; therefore, the free-field deviation (in dB) calculated also depends on the source position, measuring direction and distance from the source, and the frequency used in the test, besides other factors such as the dimensions of the room and sound absorption characteristics of all the boundary surfaces. For a broadband noise, the calculation becomes simpler since the phase factor can be ignored, and a general equation of such a calculation is given for any source position and different combinations of boundary absorption. The results of the calculation can be shown in a diagram with the aid of a microcomputer, and, hence, a three-dimensional free-field scope would be very useful as a guideline to design and to use an anechoic room.

Contributed Papers

9:35

T2. A graphic representation of acoustics using ray tracing. Philip R. Thompson (Department of Architecture, Massachusetts Institute of Technology, Cambridge, MA 02139)

This research details the development of a computer model that displays the propagation and reverberation of sound within an architectural enclosure. The visual rendering of acoustics is done by simulating sound fields using graphical ray methods. Unlike many other acoustical ray-tracing methods, in this concept, rays are emitted from a receiving point and trace the paths to the sound source so as to produce a picture of what a listener hears. This three-dimensional recursive ray-tracing algorithm has been combined with an acoustical model, utilizing specular and first-order diffuse reflections, to produce an image (or series of images) depicting the locations of real and virtual sources, the time delays in which they arrive, and their approximate energy levels. The free-path lengths of the rays are sorted into frames that produce an animation sequence demonstrating the propagation of a sound field towards the listener. The sources' emitted energy and subsequent attenuations along a ray's path result in an intensity, which is then depicted as a color at a pixel on the computer's monitor to compose an image. While the program is meant to run on most any machine, a workstation is preferred since a relatively large amount of computation (involving hundreds of thousands of rays) is required. The purpose of this analysis tool is to elucidate and verify our perceptions of room acoustics through the most readily understandable visual means.

[Work supported by MIT Project Athena through the Computer Resource Lab in the School of Architecture and Planning and by the Robert B. Newman Memorial Fellowship.]

9:50

T3. Effect of fluid-structural coupling on sound waves in an enclosure in low-frequency range. Pan Jie and D. A. Bies (University of Adelaide, Department of Mechanical Engineering, GPO Box No. 498, Adelaide, S.A. 5001, Australia)

The acoustic properties of individual normal modes in a rectangular panel-cavity system were investigated experimentally. The system consisted a heavy five-sided concrete box closed on the sixth side with a test panel. Experimental evidence in this paper reveals the dependence of the modal decay time of sound waves in the enclosure upon the modal characteristics of the test panel, such as the modal coupling factors, resonance frequencies, and modal decay times of the test panel. The results of experimental measurements are used to verify some of the previous theoretical predictions. The agreement between the measured and calculated acoustical modal decay time shows that the modal coupling theory can be used to evaluate the sound wave behavior in those enclosures, in which the coupling between fluid borne and structure borne acoustical waves is modal and where the classical model based on the locally reactive boundary assumption does not apply. The phenomenon of the maximum sound energy absorption by the panel was also observed in the experiment.

T4. Effect of fluidal-structural coupling on the acoustical decays in a reverberation room in high-frequency range. Pan Jie and D. A. Bies (University of Adelaide, Department of Mechanical Engineering, GPO Box No. 498, Adelaide, S.A. 5001, Australia)

The effect of the coupling between a sound field and test panels upon their decay behavior is investigated in high-frequency range. Quasitransient solutions of the average energies in the room and the panels are used to estimate the reverberation times in the room and to calculate the decay curves of a test panel. Experiments are conducted to verify these solutions. The estimated and measured results are in good agreement in high-frequency range, which indicates that the average decay behavior of the room and panels can be analyzed in terms of the average modal parameters. The amalgamation of the average modal parameters (of the test panels and the reverberation room) can be related to the Sabine absorption coefficient α_{Sab} of the panels. This result provides a possible interpretation for the α_{Sab} of a modally reactive surface and suggests a way to estimate this quantity by statistical energy analysis (SEA) method. The dependence of the α_{Sab} upon the room properties, in which the panel is measured, is demonstrated both experimentally and analytically.

10:45

T6. Acoustical considerations in design of semienclosed small arms training facility. Felix Sachs, Nelson Lewis, and George Luz (Bio-Acoustics Division, U.S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5422)

Semienclosed small arms training ranges combine bullet trap features of an indoor range with the ventilation of outdoor ranges. Acoustic reverberation of indoor ranges and directivity effects of outdoor ranges are also combined in the semienclosed range. When designers of a semienclosed range failed to design for the acoustic directivity of guns, community noise levels increased and residents who had adapted to a long-standing noise environment from an outdoor range began complaining. Personnel on the firing line also noticed an increase. Measurements at the firing line demonstrated a sevenfold increase in the occupational noise dose, while measurements made out to 500 m demonstrated an 8-dB increase in the community noise exposure. The 8-dB increase was attributed to the reflection of the forward weapon report from the bullet trap baffles. The increase in noise exposure at the firing line was attributed to reverberation from the side walls and roof of the semienclosed range. The measures to reduce both the occupational and community exposures are addressed.

11:00

T7. Acoustical design for the Merkin Concert Hall and the Hebrew Arts School. Peter J. George (Peter George Associates, Inc., 34 West 17th Street, New York, NY 10011)

Sound control issues were critical in the design of the multistory building that required relatively light-weight construction. The Anne Goodman Recital Hall was located directly above the Merkin Concert Hall. Dance studios were adjacent to Piano Classrooms, which were adjacent to Practice Rooms. Comparison is made between the techniques used on this building and those for the neighboring Juilliard School.

11:15

T8. Acoustical design for the Tisch School of the Arts. Peter J. George (Peter George Associates, Inc., 34 West 17th Street, New York, NY 10011)

New York University's Tisch School of the Arts has moved into a 12-story building that was gutted and recycled to provide a new home for the

T5. An indirect boundary integral equation method for architectural acoustics. D. K. Holger (Department of Engineering Science and Mechanics, 2019 Black Engineering Building, Iowa State University, Ames, IA 50011)

A numerical method and the resulting computer program for prediction of interior acoustic fields are described. The method is based upon an indirect boundary integral equation formulation of the Helmholtz equation using hybrid layer potentials. The computer program incorporates general boundary conditions directly, and utilizes piecewise constant variable approximations. It is capable of predicting the steady-state response of enclosures with finite impedance boundaries, vibrating boundaries, and multiple interior sources. Numerical results are presented for the acoustic field inside a rectangular prism under various boundary conditions and with various excitations. Apparent limitations of the current program are discussed, and various potential applications in architectural acoustics and noise control are suggested. [Work performed at the Institutt for Teleteknikk, Norges Tekniske Hogskole, Trondheim, Norway and supported by NTNf.]

10:35-10:45

Break

School. In addition to classrooms, the building includes screening rooms, edit rooms, a recording studio, an AM/FM radio station, two television studios, and a sound stage. The design approach is described and the completed results are analyzed.

11:30

T9. RASTI for speech isolation evaluation. Klaus Højbjerg (Bruel and Kjaer Instruments, Inc., 185 Forest Street, Marlborough, MA 01752)

In open offices, there is a demand for speech isolation. As it becomes more important to avoid surveillance from outside conference rooms, a reliable method of evaluating the speech security becomes necessary. Research has shown that transmission loss or the articulation index method does not give enough information. This is because speech intelligibility, or rather the lack of speech intelligibility, depends on more than the signal-to-noise ratio. The RASTI method takes into account the effects of reflections, reverberation, and noise. By using the low end of the RASTI scale, the lack of speech intelligibility can be determined. This paper describes the measurement, using elevated signal levels, as well as measurement results.

11:45

T10. Controlling the shape of an acoustic spectrum automatically in various spaces. Thor Carlsen and Richard J. Peppin (Scantek Inc./Norwegian Electronics a/s, 51 Monroe Street, Suite 1603, Rockville, MD 20850)

Often, when a room is excited by a specified spectrum, the resulting spectrum differs in shape from that desired. Caused by the impedance of the space and by the objects within, in the past, the change has been accounted for by the use of a spectrum shaper before the noise generator, which was manually adjusted so that the desired shape and level of the output was achieved. This paper presents a discussion of a bus-controlled spectrum shaper that, in conjunction with a computer controlled real-time analyzer, allows the shape of the desired spectrum to be achieved without operator intervention. Uses for vibration excitation will also be discussed.

Session U. Noise III: Alert and Notification Systems

T. James DuBois, Chairman

*Southern California Edison, 2131 Walnut Grove Avenue, Rosemead, California 91770***Chairman's Introduction—9:00*****Invited Papers*****9:05****U1. The Federal Emergency Management Agency's alert and notification systems review program.** Craig S. Wingo (Federal Emergency Management Agency, Washington, DC 20472)

Nuclear power plants throughout the United States are required by the Nuclear Regulatory Commission (NRC) to provide the means for early notification and clear instruction of the populace within the 10-mile radius Emergency Planning Zone (EPZ) around each plant. The regulations and the Federal Emergency Management Agency's (FEMA) technical program for reviewing and approving alert and notification systems are discussed in this paper. The FEMA program consists of an acoustical/engineering review of the system design, coupled with a telephone survey of residents of the EPZ following activation of the system to determine its effectiveness. To date, system designs have been submitted for every nuclear power plant in the nation. The engineering design review has been completed at 68 sites, and a telephone survey has been completed at 63 sites. Recently, the technical standards against which alert and notification systems are reviewed have come under scrutiny in Atomic Safety and Licensing Board litigation and FEMA has defended the adequacy of the current standards. Efforts are underway to place the Outdoor Sound Propagation Model used in the alert and notification system reviews into FEMA's Integrated Emergency Management Information System for application to civil defense programs, hazardous materials program, and other programs in and outside of FEMA.

9:30**U2. Design of alert and notification systems.** Louis C. Sutherland (Wyle Laboratories, 128 Maryland Street, El Segundo, CA 90245) and Eric Stusnick (Wyle Laboratories, 2001 Jefferson Davis Highway, Arlington, VA 22202)

Alert and notification systems are required by the Nuclear Regulatory Commission around nuclear power plants to provide a means of alerting the public within a 10-mile radius of a possible emergency condition that may require evacuation. The most common method of accomplishing this is through the use of audible signals from high-powered sirens. The basic problems involved in the design of such systems are discussed. These problems include specification and validation of the siren output signals, the prediction and validation of the siren signal as received in the community, and nonacoustic factors such as siren installation and control. General characteristics of currently available siren systems—both electromechanical and purely electronic types—are also briefly reviewed. Statistical aspects of short-term (second to second) and long-term (day to day) variations in siren signal levels received in the community are discussed, and this variation in the warning signal level is evaluated with respect to overall system effectiveness. Other means of conveying the warning, including the presentation of altering messages over automatically energized radios and siren-equipped emergency vehicles, are also briefly considered.

9:55**U3. Evaluation of siren alert notification systems.** Van M. Lee (Analysis & Computing, Inc., Hicksville, NY 11802)

This paper describes the essential elements in evaluating the effectiveness of a large outdoor fixed siren warning system, the state-of-the-art of the component evaluation processes, and the potential improvements in the future. The essential components in evaluating the coverage—the percentage of population within the planning zone alerted—consist of: siren acoustic performance (dB versus frequency versus directivity), site terrain and meteorological effects on sound propagation, population statistics (activity patterns and age distribution by household size), building attenuation effects, and activity-sound level response relationships. The lack of standard test procedures, whether laboratory or field, and data reporting formats has led to many disputes concerning omnidirectional and directional sirens. Standard methods for site qualification, siren testing, and data reporting similar to ANSI S1.34 (1980) and S1.35 (1979) are discussed in relation to their effect on the design and evaluation of siren systems. Also discussed are the effects of air absorption, wind and temperature gradients, operating characteristics (omnidirectional or rotating), and terrain effects, including Rasmussen's application of barrier diffraction and impedance ground effects. By correlating the physical siren

sound level coverage to the social response level of the alerted population, a systematic methodology is presented, which evaluates the overall effectiveness of a large outdoor siren alert system.

10:20

U4. Outdoor sirens: How do they work for people indoors? David N. Keast (HMM Associates, Inc., 336 Baker Avenue, Concord, MA 01742)

People are normally indoors where the sounds from outdoor warning sirens are greatly attenuated by building structures. Nevertheless, practical experience and the results of tests by the Federal Emergency Management Agency and others have shown that siren sounds are quite effective for alerting large populations—including those indoors. Why is this so? Based upon data in the literature, three possible explanations are examined in this paper: (1) the possibility that the sounds from sirens outdoors, even though attenuated through building structures, are still sufficient to directly alert people indoors; (2) the possibility that people indoors are more responsive to siren sounds than people outdoors; and (3) the possibility that people indoors are alerted by other people who are outdoors when they hear the siren sounds. This examination indicates that all three mechanisms could be contributing to the effectiveness of siren sounds indoors. Their relative significance is not known, but probably varies with geographic location, weather conditions, and time of day.

Contributed Paper

10:45

U5. The influence of high-intensity sounds on the performance of air warning sirens. Frederic G. Pla (Sverdrup Technology, Inc., NASA LeRC Group, 16530 Commerce Court, P.O. Box 30650, Midpark Branch, Middleburg Heights, OH 44130)

The use of sirens as air warning devices has been widespread for several hundred years. Due to the very high sound-pressure levels generated, finite-amplitude effects can result in large losses leading to poor sound generation efficiency. Although much work has been done on siren de-

sign, the importance of finite-amplitude effects has been neglected in the recent past, even though a review of earlier works shows that researchers such as Reyleigh, in 1903, King, in 1919, and later, Hart, in 1926, noticed the extra acoustic losses associated with high-intensity sound sources. Comparisons between linear and nonlinear models of wave propagation in a siren horn illustrate the importance of these types of losses, and show how they can be minimized. Acoustic saturation, which limits the sound-pressure level that can be perceived by a receiver at a given distance from a high-intensity sound source, and the implications of finite-amplitude effects on the rating of high-intensity sound sources in a laboratory environment are also discussed.

WEDNESDAY MORNING, 18 NOVEMBER 1987

TUTTLE CENTER ROOM, 9:00 TO 11:50 A.M.

Session V. Physical Acoustics III: Scattering, Diffraction, and Radiation

Ronald A. Roy, Chairman

Department of Mechanical Engineering, Mason Laboratory, Yale University, 9 Hillhouse Avenue, New Haven, Connecticut 06520

Chairman's Introduction—9:00

Contributed Papers

9:05

V1. Monostatic angular distributions and their relation to resonances in acoustical scattering from fluid loaded solid elastic objects. G. Gaunaud (NSWC, White Oak, Silver Spring, MD 20903-5000) and M. F. Werby (NORDA, NSTL, MS 39529)

It is possible to detect resonances from backscattered echoes (form functions) by varying the nondimensional frequency $kL/2$ in small increments. This gives rise to a familiar pattern of nulls followed by a sharp rise for the spherical case and similar results for the spheroidal case. This

approach, however, limits one to one aspect angle for three-dimensional nonspherical problems and often entails tedious and time consuming calculations to find the often narrow resonance regions. Another approach suitable for three-dimensional problems is to examine closely spaced monostatic angular distributions enabling one to look for possible resonances over a continuous range of aspect angles. This method is used to examine scattering from solid steel spheroidal shells. The results indicate that Rayleigh type resonances are aspect angle dependent in contrast to our recent results on spheroidal shells. The results are consistent with the standing wave interpretation of resonances proposed by use earlier for shells and employed in work on material properties of solid elastic spheroids.

V2. Acoustic scattering from an elastic prolate spheroid in a shallow-water waveguide. Gary S. Sammelmann and Roger H. Hackman (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

The bistatic scattering of a broadband acoustic pulse from an elastic prolate spheroid in a shallow-water waveguide is studied. The calculation is based on the formal solution to the scattering in an inhomogeneous-layered waveguide presented at a previous meeting of the Acoustical Society [R. H. Hackman and G. S. Sammelmann, *J. Acoust. Soc. Am. Suppl. 1* **81**, S42 (1987)]. The waveguide is assumed to be bounded from above by a sound soft air-seawater interface, and from below, by a rigid sediment-seawater interface. The sound speed is taken to be a monotonically decreasing function of depth. The emphasis of this study is on extracting elastic information from the scattered waveform. The importance of the rescattering of the acoustic pulse between the boundaries and the target is also studied.

V3. Acoustic scattering from large aspect ratio, axisymmetric shells. Roger H. Hackman and Douglas G. Todoroff (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

The spheroidal-coordinate-based transition matrix has been successfully applied to the solution and the analysis of the acoustic scattering from large aspect ratio solids [R. H. Hackman and D. G. Todoroff, *J. Acoust. Soc. Am.* **78**, 1058-1071 (1985)]. However, this approach required special care due to the existence of elastic states that were weakly coupled to the external acoustic field. Here, a new version of the spheroidal formalism is presented, one with improved stability and which is better suited to thin shells. This approach is applied to the acoustic scattering from a large aspect ratio, elastic prolate spheroidal shell of constant thickness. Theoretical predictions are compared with experimental measurements, and a physical explanation is given for the more prominent features in the backscattered formfunction.

V4. Comparison of quadratic and piecewise-constant boundary element solutions of the surface Helmholtz equation. Joseph J. Shirron (Sachs/Freeman Associates, Landover, MD 20785)

A comparison of the accuracy and computation time of two different boundary element solution techniques for the surface Helmholtz equation of a rigid scatterer is presented. In the simplest case, the body's surface is represented by an assemblage of planar triangles, over which the surface pressure is taken to be constant. In the more sophisticated case, quadratic-isoparametric elements are used. Results are presented for the comparison of rates of convergence with mesh size and CPU times for the two methods. The effects of varying the order of Gaussian quadrature are shown. Further results are given comparing the efficiency of Schenck's algorithm for eliminating the nonuniqueness problem at the body's interior eigenvalues with the modified Green's function technique of D. S. Jones. The quadratic-isoparametric solution method together with Schenck's algorithm is found to be the more efficient means of obtaining an approximate solution to the surface Helmholtz equation. [Work supported by Naval Research Laboratory, Washington, DC.]

V5. Another look at the Rayleigh angle backscattering. Peter B. Nagy and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

The increased backscattering from a liquid-solid interface at the Rayleigh angle is a well-known indication of leaky surface wave generation (similar phenomena were observed from thin plates immersed in liquid

due to leaky Lamb wave generation, too). This simple technique has proved to be very useful in material evaluation via measuring both surface and Lamb wave velocities. On the other hand, a comprehensive experimental study [M. de Billy, L. Adler, and G. Quentin, *J. Acoust. Soc. Am.* **72**, 1018 (1982)] showed much stronger backscattered signals than predicted later by theoretical works based on back reflection. New experimental results are presented demonstrating that this back reflection is negligible in the backward signal, which is simply backscattering from inevitable material inhomogeneities greatly enhanced by surface or Lamb wave generation. Therefore, beside measuring macroscopic elastic properties in the previously suggested way, the backscattered signal can be used to characterize surface and near-surface material inhomogeneities as well. [This work was supported by the U. S. Department of Energy, Basic Energy Sciences, Grant DE-FG02-84ER45057.A004.]

V6. Measurements of Rayleigh wave scattering at a corner. Peter B. Nagy and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

The problem of Rayleigh wave scattering in a homogeneous, isotropic wedge by experimental means was investigated. For an incident Rayleigh wave, reflected and transmitted Rayleigh waves are generated, as well as longitudinal and transverse cylindrical bulk waves initiated by the virtual source at the corner line. Sector-shaped test pieces were used, which focused back the diverging bulk waves to the same corner line after normal reflection at the cylindrical surface. By this simple geometry, both the phase and amplitude of all four components were able to be measured. Experimental results are presented to show the varying energy partition among these modes as a function of wedge angle from 90° to 150° in aluminum. Furthermore, the Rayleigh wave reflection and transmission results are compared to recent theoretical predictions by Gautesen [*Wave Motion* **9**, 51-59 (1987)]. [This work was supported by the U. S. Department of Energy, Basic Energy Sciences, Grant DE-FG02-84ER45057.A004.]

V7. Farfield patterns of rigid nonaxially symmetric obstacles: Example calculations using the Waterman scheme for approximation of the T matrix. Angie Sarkissian (Sachs/Freeman Associates, Landover, MD 20785) and Allan G. Dallas (Code 5131, Naval Research Laboratory, Washington, DC 20375-5000)

To our knowledge, there have been no reports of numerical applications of the Waterman scheme to rigid scatterers of nonaxially symmetric geometry. Here, the results of computations for various slender obstacles of this sort are described. Emphasis is placed upon the amount of work required for such calculations (relative to that necessary for axially symmetric shapes) and upon specific details, such as the numbers of spherical waves and quadrature points required.

V8. The effect of finite surface acoustic impedance on sound fields near a smooth diffracting ridge. Ji-xun Zhou, James A. Kearns, Yves H. Berthelot, and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Laboratory scale experiments using spark sources are being conducted to study long-range propagation of sound over hills and valleys. The effect of the finite impedance of the diffracting scaled hill on the acoustic field is investigated, with either plywood or plywood covered with carpet. Particular attention is given to the case of small grazing angles of incidence of the sound wave on the diffracting surface. The model of Delany and Bazley is used to determine the impedance of the material from the value of the effective flow resistivity σ . The insertion loss of the scaled ridge is inferred from measurements of the ratio of the Fourier transforms of the diffracted pressure field to that of the free field, with subsequent

digital signal processing performed in the frequency domain. Experimental data are compared with the theory of matched asymptotic expansions [J. Acoust. Soc. Am. Suppl. 1 79, S30 (1986)] for situations where the diffracted acoustic pressure is measured either along the ridge, or along a vertical axis at the apex or behind the ridge. [Work supported by NASA, Langley Research Center.]

11:05

V9. Diffraction of pulses at a sharp edge. Gerrit Schouten (Aerospace Department, Delft University of Technology, Kluyverweg 1, 2629 HS Delft, The Netherlands)

The closed expression for the diffracted field of a sound pulse is derived in a straightforward manner using Sommerfeld's method for the generation of many-valued solutions from known one-valued solutions of the wave equation. The spherical primary wave front carries a delta-function disturbance, whereas the spindle-shaped diffraction front carries a square-root singularity bounding the filled diffraction region where two dimensional aspects are dominating. Integration w.r.t. time of firing of the pulse yields the transient behavior of a hydrodynamic switch-on source with a unit-step wave front. The well-known result of Cagniard (1935) is thus reproduced in a direct way. Plots of the velocity field inside the diffraction region show the peculiarities of the transition from the wave solution to the steady Laplace solution. Applying the method of descent to the three-dimensional solutions yields the known two-dimensional equivalents, again in a more direct way than known from the literature [R. D. Turner (1956); H. A. Lauwerier (1962); E. M. de Jager (1964)].

11:20

V10. Acoustic power radiation by cylindrical stiffened shells. B. Laulagnet and J. L. Guyader (Laboratoire Vibrations-Acoustique, Institut National des Sciences Appliquées de Lyon, 69621 Villeurbanne Cedex, France)

Sound radiation by finite cylindrical stiffened shells, in light or heavy fluid, is studied using modal calculation. Numerical results on power radiation, shell quadratic velocity, and radiation factor are given, and the influence of parameters like stiffeners distribution, stiffener and shell damping, type of excitation, etc., is discussed. Finally the analysis of shell modes that contribute most to the radiation is presented. It is shown, in particular, that modes whose structural losses equal radiation losses dominate the power radiation, whatever the fluid (however, for light fluids, these modes have high radiation efficiency and low radiation efficiency for heavy fluids). This analysis allows the understanding of radiation curve singularities, and shows that critical frequency phenomena are completely different for heavy and light fluids. [Work supported by the Ministère de la Défense Contrat D.R.E.T.N° 85-065.]

11:35

V11. On the acoustic excitations of an elongated elastic solid. Kevin L. Williams, Douglas G. Todoroff, and Roger H. Hackman (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407)

Previously, an analysis of the acoustically coupled, axisymmetric, elastic excitations of a fluid-loaded, prolate spheroidal target was presented [D. H. Trivett, G. S. Sammelmann, and R. H. Hackman, J. Acoust. Soc. Am. Suppl. 1 81, S13 (1987)]. At this meeting, a controversy arose as to the phase speed of the elastic excitations. Here, direct experimental and theoretical evidence is presented that supports our contention that the velocity of propagation of the elastic waves that propagate on an elongated solid target at low frequencies is essentially the bar speed.

WEDNESDAY MORNING, 18 NOVEMBER 1987

TUTTLE NORTH ROOM, 9:00 TO 11:50 A.M.

Session W. Structural Acoustics and Vibration III: Analysis and Measurement of Structural Vibrations

Joel M. Garrelick, Chairman

Cambridge Acoustical Associates, Inc., 54 Rindge Avenue Extension, Cambridge, Massachusetts 02140

Chairman's Introduction—9:00

Contributed Papers

9:05

W1. Impact testing of light objects. Mohamed K. Abdelhamid (Production Engineering Department, Helwan University, Helwan, Cairo, Egypt) and Anna L. Pate (Department of Engineering Science and Mechanics, Iowa State University, Ames, IA 50011)

This paper investigates the design of a measurement system that measures the impact signature of a light object. The design is based on a freely supported piezoelectric force transducer. Effects of top mass, base mass, and stiffness of the transducer were investigated. Two models were used in the analysis of the impact, a linear model with a 3 degree of freedom system and a Hertzian model. Using the Hertzian model, a criterion for choosing the optimum size of the base mass was developed. This criterion is based on the least-square error in the force spectrum. Similar results were obtained by the linear model and numerical simulations. Experimental confirmation of the results was conducted using small Teflon and ny-

lon balls and a piezoelectric force transducer. The results agreed well with the model predictions.

9:20

W2. Wave effect correction to reciprocity calibration results of vibration primary standards. Li-Feng Ge (National Bureau of Standards, Gaithersburg, MD 20899 and Anhi Bureau of Standards and Metrology, Hefei, People's Republic of China)

The paper reveals the stress wave effect (SWE) on reciprocity calibration results of vibration primary standards and derives exact correct formulas of mechanical standards according to reciprocity theory and stress wave theory. The paper further expounds that both the electrodynamic reciprocity method (ERM) and the piezoelectric reciprocity method

(PRM) need this correction at higher frequencies and shows that taking this measure can eliminate a system error of 10^{-2} to 10^{-3} orders of magnitude.

9:35

W3. Viscoelastic effects on solid amplitude and phase transformers. L. M. B. C. Campos (Instituto Superior Técnico, 1096 Lisboa Codex, Portugal)

The longitudinal oscillations of nonuniform bars are used as displacement amplifiers in power tools and other devices. These often operate near resonance [E. Eisner, *J. Acoust. Soc. Am.* **35**, 1367-1372 (1963)], so that the elastic model should be replaced by a viscoelastic one, which includes damping. In the present paper, the longitudinal vibrations of a tapered viscoelastic bar are discussed generally from the wave equations for the displacement and strain. Exact solutions are obtained for the exponential, catenoidal, sinusoidal, and inverse shapes, and also for the Gaussian and power-law shapes. These solutions generalize earlier results for elastic bars; e.g., the elastic Gaussian bar [D. A. Bies, *J. Acoust. Soc. Am.* **34**, 1567-1569 (1962)] is generalized to a viscoelastic one. Diagrams of wavenumber and damping ratio versus frequency and viscous relaxation time are presented for several shapes of tapered bar; they describe the propagation and dissipation of oscillations and their effects on amplitude and phase.

9:50

W4. Relationship between the impedances and the reflection and transmission coefficients of a junction. L. J. Maga and G. Maidanik (David Taylor Naval Ship Research and Development Center, Bethesda, MD 20084)

A junction constitutes the coupling between dynamic systems. In this case, the one-dimensional dynamic systems so coupled are characterized by their wavenumbers and mass densities. The interface impedances of the dynamic systems at the junction may then be defined. The junction may be defined in terms of a junction impedance matrix. The nature of this matrix will be discussed. This matrix, together with the interface impedances of the dynamic systems, may then be employed to derive the junction interaction matrix, which consists of reflection and transmission coefficients. The manner and results of this derivation will be discussed and a few examples will be cited.

10:05

W5. Damping and vibration analysis of bonded beams with a lap joint. Mohan D. Rao and Malcolm J. Crocker (Mechanical Engineering Department, Auburn University, Auburn, AL 36849)

In this paper, a theoretical model to calculate the loss factors and the resonance frequencies of flexural vibrations of a system of two parallel beams bonded with a single lap joint by a viscoelastic material has been developed. First, equations of motion of the joint region are derived using a differential element approach considering the transverse displacements of the upper and the lower beam to be different. The normal force between each beam and the adhesive layer is represented by a Pasternak base model, which consists of closely spaced linear springs. The shear force at the interface is modeled using a viscous model for friction. The resulting equations of motion, together with equations of transverse vibrations of the beams away from the joint, are solved using motion continuity conditions and boundary conditions at the free ends of the beams. Equations for calculating the resonance frequency and the loss factor for the case of clamped-clamped boundary conditions are then derived. Numerical results are generated and are compared with experiments for a system of two beams with a simple lap joint made of graphite epoxy composite material. [Work supported by NASA-MSFC.]

10:20

W6. Vibration analysis of mass-loaded beams. Dhanesh N. Manikanahally and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

A general procedure for determining the dynamic response of a mass-loaded free free beam subjected to a harmonic and transient force is given. Though free free beams are considered for analysis, the same procedure could be extended for other end conditions also. The beam is assumed to have structural damping for determining the steady-state response due to harmonic force excitation. The mode shapes for free vibration, dynamic response, and dynamic strain due to forced excitation are presented in a graphical form. The analysis is used to study a space structure, modeled as a mass-loaded free free beam, by making an exhaustive optimization search for minimum dynamic response due to harmonic and transient excitation forces. The computer program developed for the analysis is used to check some simple beam problems [G. B. Warburton, *The Dynamical Behaviour of Structures* (Pergamon, New York, 1976)]. [Work supported by SDIO/DNA, Contract No. DNA-001-85-C-0183.]

10:35

W7. Finite element analysis of a large vibrating space structure. B. S. Sridhara and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

The large flexible space structure to be used in the Strategic Defense Initiative Project is modeled as a long flexible beam with three point masses. Using the Galerkin method, finite element equations are formulated that take advantage of the fact that the natural boundary conditions come out as a result of integration by parts. Hermite polynomials have been chosen for the shape or interpolation function. A tubular beam made of graphite epoxy material, 100 m in length with 2.0-m o.d. and 1.95-m i.d. is considered for the purpose of this analysis. Masses of 500, 10 000, and 1000 kg are placed at the left extreme end, at a distance of 20 m from the left extreme end, and at the right extreme end of the beam, respectively. Natural frequencies of the long flexible beam are calculated using standard methods and the mode shapes are also plotted. Results are also obtained for a different location and increased values of the largest mass. Computer codes are being prepared to obtain the dynamic and steady-state response of the beam subjected to an impulse and a sinusoidal force.

10:50

W8. Procedure for the measurement of wavenumber/frequency admittance of structures. Karl Grosh, W. Jack Hughes, and Courtney B. Burroughs (Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

A procedure for measuring the wavenumber/frequency admittance of structures using an array of drives in a standing wave pattern was developed in a paper presented in the ASA meetings in Indianapolis, IN on 14 May 1987 [K. Grosh, J. H. Hughes, and C. B. Burroughs, *J. Acoust. Soc. Am. Suppl.* **1** **81**, S73 (1987)]. In this paper, data are presented to verify the validity of the measurement procedure. The wavenumber/frequency spectra are presented for arrays of point drives and the vibration responses of lightly damped beams and long beams with heavily damped ends. The effect of steering the array to different wavenumbers and varying the number of drives is examined.

11:05

W9. Influence of various parameters on the transmission of vibrational power. T. Gilbert, J. M. Cuschieri, M. McCollum, and J.L. Rassinoux (Center for Acoustics and Vibrations, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The transmission of vibrational power between two thin plates in an L-shape configuration is investigated using an SEA model. Only bending waves are considered in the model. Expressions are developed for both the ratio of energy levels in the two plates and the ratio of the transmitted power to the input power. It is found that these ratios are only dependent upon three parameters: frequency, dampings of the plates, and coupling loss factors between the two plates. Therefore, a way to decrease both the

transmitted power and the energy level in the receiving plate over a wide range of frequencies is to increase the damping of the source plate. The coupling loss factor is influenced by the physical characteristics of the plates, such as area, thickness, and material. The effect of these parameters on the coupling loss factor and on the power and the energy ratios is found to be small. Thus the most efficient way to control the transmission of power and the energy levels is to increase the damping of the plates. This result is in agreement with other results obtained using the energy influence coefficient method. [Work supported by NASA.]

11:20

W10. Power flow method for an L-shaped plate. J. L. Rassinoux, J. M. Cuschieri, M. McCollum, and T. Gilbert (Center for Acoustics and Vibrations, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The power flow method is extended to the analysis of the vibrational power flow for two-dimensional L-shaped plate structures. The analysis is for a plate that is assumed to be simply supported along the edges and pinned at the junction. Structural damping is introduced by means of a complex elasticity modulus. The transmitted power from one plate to the other is computed using input and transfer generalized mobilities. The mobility functions are obtained from the solution for the response of a flat plate using either point or edge moment excitation. The analytical solution obtained enables computation of the response for a broadband excitation, regardless of the range of frequency considered. Different locations of the excitation point and different values of damping and their influence

on the response are analyzed. As a verification, the obtained results are compared with finite element analysis and statistical element analysis. [Work supported by NASA.]

11:35

W11. Experimental measurement of power transmission in an L-shaped beam. M. McCollum, T. Gilbert, J. Rassinoux, and J. Cuschieri (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The vibrational power transmitted across the joint between two beams forming an L-shaped structure, when one of the beams is excited by a point force at the free end, is measured experimentally. One method of calculating the power transmission is based on the assumption that the power transmitted to the receiver beam is equal to the power dissipated by that beam. The damping loss factor used in the calculation is obtained from measurements on the receiver beam. The results of the experimental analysis are compared to those obtained using a power flow technique, the results of which have been verified by the closed form solution for the global structure. The measurements show very good agreement with the analytical results in terms of the location of the resonant frequencies, except for low frequencies. The measured levels are, however, significantly lower than the predicted levels, especially at the resonant peaks, indicating that the measured damping levels are underestimated. The measured levels are generally very close to the predicted levels away from the natural frequencies. [Work supported by NASA and ONR.]

Session X. Education in Acoustics I: Project Laboratories for Undergraduates

Allan D. Pierce, Chairman

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

12:30

X1. Development of an undergraduate experimental acoustics center. Jacek Jarzynski (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A center for experimental acoustics and ultrasonics was recently developed for use in a senior-level laboratory course that is required for the BSME degree at Georgia Tech. The center is a distinct facility with permanently assigned instrumentation and equipment available for hands-on performance of experiments and laboratory projects by students; possible experiments are related to current course offerings and on-going research activities within the School of Mechanical Engineering. The experiments that have been developed for the center allow the students to study some of the basic features of wave motion and to learn about some typical applications of acoustics. Representative experimental tasks that can be carried out within the center include: measurements of pressure levels and frequency spectra of sound from environmental sources; application of the ultrasound to nondestructive testing of materials; and measurement of fluid flow using an ultrasound Doppler technique. To the fullest extent possible, the instrumentation of the center has been chosen such that the students will be introduced to the latest state-of-the-art data acquisition, storage, and processing techniques. The objectives of this type of laboratory instruction are outlined, with some sample student comments. The illustrated talk will be accompanied by several live demonstrations of the experiments that students carry out within the center.

1:10

X2. Project-type acoustics experiments in the USNA acoustics lab. S. A. Elder (Physics Department, United States Naval Academy, Annapolis, MD 21402)

For some years, the Naval Academy Physics Department has offered a senior-level course in physical acoustics, accompanied by a weekly laboratory. Since the classes are small, the laboratory offers a chance for tutoring the students in more advanced and creative projects than the usual "classic" lab experiments for courses at this level. Usually there is time to perform three or four projects over the semester, each one defined by a general area of interest (such as environmental acoustics, architectural acoustics, computational and data acquisition methods, etc.). Sometimes the project spills over into the next semester and becomes a credit earning research project beyond the course itself. Examples of successful student projects are: construction and testing of a loudspeaker (magneplanar, flame driven, etc.), acoustical investigation of a local concert hall or church, construction and testing of an acoustic guitar, development of computer-controlled data acquisition and analysis system. Specific examples, as well the general philosophy of the course, will be given.

1:30

X3. Sonar project laboratory for undergraduates. Murray S. Korman (Department of Physics, United States Naval Academy, Annapolis, MD 21402)

A portable microcomputer workstation for supporting basic acoustic experiments has been developed over the past three years in the Physics Department at the U.S.N.A. Students enrolled in underwater acoustics and sonar (a 3-h lecture course without a laboratory) perform fundamental acoustics experiments in the classroom without having to utilize large numbers of analog electrical boxes. The heart of the workstation (based on the 6502 microprocessor) includes a low-cost, dual-channel, 8-bit analog-to-digital board. This workstation features menued software that can capture a waveform and display its trace. Simple mathematical operations include squaring the waveform, integrating, differentiating, finding the rms value, and determining the probability density function. A 1024-point FFT program is also included in the menu. Students are not required to know computer languages to operate the workstation and no experiment is automated or simulated. During nonclass hours, students design an experimental project that might include "model" measurements of target strength, beam pattern functions, reflection loss, sound velocity, vibration, and studies of signals in noise. The workstations's versatility will be demonstrated.

Session Y. Speech Communication IV: Analysis and Coding

Stephen A. Zahorian, Chairman

*Department of Electrical Engineering, Old Dominion University, Norfolk, Virginia 23508***Contributed Papers**

12:45

Y1. Vector quantization of LPC parameters based on dynamical features of hearing. Shigeru Ono and Takashi Araseki (NEC Corporation, C&C Information Technology Research Laboratories, Kawasaki 213, Japan)

New code-word search criteria for vector quantization of LPC parameters are proposed. Various vector quantization schemes for LPC parameters have been proposed previously. In these schemes, an input speech waveform is partitioned into successive short time frames and code-word search is carried out under the criteria based on instantaneous speech features. Characteristics of quantization noise vary widely from frame to frame, independent of the dynamic characteristics of the input speech. At very low bit rates, where it is difficult to keep the quantization distortion at a low level, variation in the quantization noise significantly degrades the quality of the reconstructed speech. "Temporal masking" experiments indicate that there is a positive correlation between the masking value and the speech spectral transition rate. These experiments also suggest that spectral transition rate distortion is emphatically perceived. In this paper, new criteria, based on these temporal masking properties, are proposed. They would minimize not only the instantaneous spectral distortion, but also the spectral transition rate distortion. Using these criteria, vector quantization of the LPC parameters is developed. Subjective experiments confirm that these criteria are superior to those based only on instantaneous speech features.

1:00

Y2. LPC voicing periodicity correction. Paul Milenkovic and Sotaro Sekimoto (Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53705-2280)

The autocorrelation method of linear predictive coding (LPC) analysis matches the first p samples of the autocorrelation function of the impulse response of an all-pole filter to the autocorrelation function computed from a speech waveform. The autocorrelation function of the speech waveform is influenced by the periodicity of the voice source, as well as by the waveform window. This influence, when incorporated into the autocorrelation function of the all-pole filter, results in errors in formant frequency and bandwidth values that become important at the fundamental frequencies typical of female speech. The use of an analysis-by-synthesis technique is proposed, where the autocorrelation function of the output waveform of an LPC synthesizer is matched to the speech data autocorrelation function. The all-pole filter determined by LPC analysis provides a match when the synthesizer is operated at an extremely low fundamental frequency. The synthesizer fundamental frequency is increased in small steps and corrections are made to the all-pole filter at each step to preserve the autocorrelation function match. This process continues until the correct value of the fundamental frequency is reached. [Work supported by NIH Grant NS 21516-3.]

1:15

Y3. ARMA speech analysis as a means for studying articulation. Louis Boves and Johan de Veth (Institute of Phonetics, Nijmegen University, P.O. Box 9103, 6500 HD Nijmegen, The Netherlands)

It is well documented that linear prediction analysis will only allow prediction coefficients to relate to vocal tract shapes under very special conditions, and that the problems in interpreting LP analysis results in

articulatory terms are due to the simplifying assumptions underlying the analysis model. Usually, the assumption that the system is all pole [or auto-regressive (AR)] is thought to be the most important simplification. Therefore, it is hoped that the use of the less restrictive pole-zero [or auto-regressive moving average (ARMA)] analysis model will offer greater opportunities for an articulatory interpretation. A detailed study of a number of different ARMA analysis implementations has shown that the all-pole assumption may not be the most important restriction on the articulatory interpretation of the analysis results; the assumption of Gaussian excitation may prove to be at least equally impeding. [Research supported by the Foundation for Speech Technology, funded by SPIN.]

1:30

Y4. Evaluation of articulatory codebooks. J. N. Larar, J. Schroeter, and M. M. Sondhi (AT&T Bell Laboratories, Murray Hill, NJ 07974)

The design of a codebook of articulatory shapes was recently reported [M. M. Sondhi, J. Schroeter, and J. N. Larar, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S78 (1987)]. The codebook is used in an automatic articulatory analysis/synthesis scheme relying on good initial estimates of vocal tract cross-sectional areas to start up an optimization procedure [J. Schroeter, J. N. Larar, and M. M. Sondhi, *Proc. IEEE ICASSP*, 308-311 (1987)]. An application-oriented way to evaluate the codebook would be to determine how much the final optimized shape deviates from the shape initially selected. A more general evaluation is to compute the average distortion incurred in quantizing some input record of speech. Using this procedure with an appropriate distance measure, various aspects of codebook formation are evaluated. In particular, the focus is on the strategies utilized for generating training data. Also considered is the effect of different approaches to code-word formation once the training data have been obtained. Evaluation results for this work done to refine the codebook will be presented along with the performance as a function of codebook size.

1:45

Y5. Speaker adaptation in articulatory speech analysis by synthesis. J. Schroeter, J. N. Larar, and M. M. Sondhi (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Using an articulatory approach for low-bit rate speech coding [J. Schroeter, J. Larar, and M. M. Sondhi, *J. Acoust. Soc. Am. Suppl.* 1 **80**, S19 (1986); *Proc. IEEE ICASSP* 308-311 (1987)], means for adapting an analysis/synthesis scheme to different speakers was investigated. Since a crucial feature for this approach to speech coding is the use of an articulatory codebook of tract shapes and related transfer functions [M. M. Sondhi, J. Schroeter, and J. N. Larar, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S78 (1987); J. N. Larar, J. Schroeter, and M. M. Sondhi, *J. Acoust. Soc. Am. Suppl.* 1 **82**, S54 (1987)], the codebook entries have to be modified to accommodate different nominal vocal-tract sizes. Two different ways of speaker adaptation are compared. In the first approach, the codebook is comprised of several subcodebooks, each for a different size of the vocal tract. In this case, a method for determining the optimal subcodebook for each speaker is needed. Alternatively, in the second approach, the LPC-representation of the original speech is scaled for access to a constant tract-size codebook. Then, the codebook entry found is inversely scaled for synthesis. Here, a method for determining the optimal scale factor is required.

2:00

Y6. Speech formant trajectory estimation using dynamic programming with modulated transition costs. David Talkin (AT&T Bell Laboratories, Room 2D-410, 600 Mountain Avenue, Murray Hill, NJ 07974)

A new algorithm to track automatically speech formant frequencies have been developed. Dynamic programming is used to optimize formant trajectory estimates by imposing appropriate frequency continuity constraints. The continuity constraints are modulated by a stationarity function. The formant frequencies are selected from candidates proposed by solving for the roots of the linear predictor polynomial computed periodically from the speech waveform. The local costs of all possible mappings of the complex roots to formant frequencies are computed at each frame based on the frequencies and bandwidths of the component formants for each mapping. The cost of connecting each of these mappings with each of the mappings in the previous frame is then minimized using a modified Viterbi algorithm. Two sentences spoken by 88 males and 43 females were analyzed. The first three formants were tracked correctly in all sonorant regions in over 80% of the sentences. These performance results are based on spectrographic analysis and informal listening to formant-synthesized speech.

2:15

Y7. Some properties of autoregressive model related cepstrum. Bing-Hwang Juang and David Mansour (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Autoregressive model related cepstrum, or, briefly, LPC cepstrum, recently gained renewed attention because it was shown to be an effective representation in speech recognizer designs [B. H. Juang, J. G. Wilpon, and L. R. Rabiner, Proc. IEEE ICASSP-86, 765-768 (1986); B. Hanson and H. Wakita, Proc. IEEE ICASSP-86, 757-760 (1986); Y. Tohkura, Proc. IEEE ICASSP-86, 761-764 (1986)]. In this paper, some properties of the LPC cepstrum are investigated that are of important consideration when the recognizer is to be deployed in situations where mismatch between the training conditions and the testing conditions may potentially occur. More specifically, the effects of LPC analysis order, additive noise, and pole movements upon some key characteristics of the LPC cepstrum are considered, such as the average and the norm of the cepstral vector. A simple relationship among the cepstral coefficients, the predictor coefficients, and the reflection coefficients is established, and unlike the Itakura-Saito distortions, it is shown that the L_2 distortion between two cepstra obtained from different-order LPC models of the same speech data may contain an inherent nonzero bias. Also considered are the effects of additive noise, using an additive power spectrum model and a constrained ARMA model (sum of an all-pole model and a constant). It is shown that the norm of the LPC-cepstral vector shrinks as a result of noise contamination, compared to that of the clean speech. Further, moving the poles of an LPC-model spectrum is shown to be equivalent to multiplying a power series to the cepstrum and thus LPC-model bandwidth broadening and pole enhancement can be easily accomplished in the cepstral domain. Consequently, evaluation of some frequency-weighted distortion measures that are believed to be desirable under certain conditions becomes straightforward via cepstral processing.

2:30

Y8. Speech enhancement in crypto-bridge communication systems. Daniel Lin and Yacov Yacobi (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

In secure audio teleconferencing, a crypto-bridge system synchronously adds simultaneous encrypted signals, modulo a known number P , and broadcasts the result to the participants. Each terminal, using a secret key, then decrypts the modular sum of encrypted signals to obtain the desired ordinary sum of plain text signals. Such secrecy systems are provably secure, assuming the existence of one-way functions (e.g., DES) for the modular sum cipher [E. F. Brickell *et al.*, CRYPTO-87]. In this paper, the implementation of additive crypto systems in a narrow-band ($2B + D$) ISDN environment are considered. The problem of noise reduction for 6-bit linear PCM speech coding (an essential constraint for

bridging the encrypted signals in a single B-voice channel) is discussed. In particular, an enhancement procedure using a prefilter, "dither" noise and an adaptive postfilter is proposed. The system is based on an all-pole model for the degraded speech. The coefficients of the postfilter are easily derivable from the all-pole filter parameters. Informal listening tests have shown that the enhanced speech is near "toll" quality.

2:45

Y9. Invariant acoustic cues in stop consonants: A cross-language study using the Wigner distribution. H. Garudadri (Electrical Engineering, University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada), J. H. V. Gilbert, A-P. Benguerel (Audiology and Speech Sciences, University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada), and M. P. Beddoes (Electrical Engineering, University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada)

Acoustic invariance was investigated in 200 unaspirated stops from English, Telugu, and French, using the Wigner distribution. The bilabials were characterized either by absence of any noticeable burst, or by an energy concentration in the low-frequency region during the burst, when present. The burst of alveolars was mostly diffuse in nature, but sometimes became compact, especially when it was followed by a back vowel. The burst of velars had a compact spectrum centered near F_2 of the vowel, when followed by back vowels, but was most often diffuse when followed by a front vowel. The burst of all dental from Telugu was found to be diffuse. When the burst of alveolars was compact, or that of velars diffuse, the duration from the start of the burst to the start of the vowel and F_2 of the vowel were found to be reliable cues for place of articulation. The diffuse nature of bilabials and alveolars, reported by Lahiri *et al.* [J. Acoust. Soc. Am. 76, 391-404 (1984)] and by the authors referenced therein, can be partly attributed to the temporal averaging of the burst spectrum and the diffuse vowel spectrum, which is inevitable in Fourier-type analyses [Garudadri *et al.*, J. Acoust. Soc. Am. Suppl. 1 79, S94 (1986)]. Unlike in other investigations, no apparent pattern could be found for the tokens wrongly classified. [Work supported, in part, by NSERC, Canada.]

3:00

Y10. A new mathematical model for intonation: Historical motivation and application. Deborah Rekart (Department of Speech Communication, The Pennsylvania State University, University Park, PA 16802), and David Drumright (Department of Mechanical Engineering, The Pennsylvania State University, University Park, PA 16802)

Early attempts to characterize English intonation were theoretical descriptions in terms of tones, tunes, and segmental pitch levels. Recent acoustic descriptions of intonation have involved the perceptual determination of F_0 contours in terms of global declination lines or as breath groups with terminal and nonterminal slopes. Lieberman *et al.* (1985) showed that F_0 contours could be described objectively by all-points linear regression lines, but this did not yield a compact characterization of the contours. Hirst (1983) developed a model based on nonlinear (parabolic) function, but it is not unified in that it resets parameters for syllables independently. Attempts to compare intonation contours in different languages and dialects have been based on the existing models. In the following abstract a mathematical model capable of compactly quantifying intonation contours using four parameters is proposed [Drumright and Rekart (1987)]. This paper demonstrates how the model can be applied to analyze foreign language, non-native, and dialectal differences in the intonation of statements and questions in spontaneous and read speech.

3:15

Y11. A new mathematical model for speech intonation: Theory and hardware/software realization. David G. Drumright (Department of Mechanical Engineering, The Pennsylvania State University, University

Park, PA 16802), and Deborah M. Rekart (Department of Speech Communication, The Pennsylvania State University, University Park, PA 16802)

A mathematical model based on combinations of sinusoids and exponential decay is shown to be superior in representing actual intonation contours. The model is generative rather than *ad hoc*, in that the configuration

is controlled by four overall parameters. The model has been realized and tested in software form by a program that reads an actual intonation curve [produced by the program PIG and described in J. Acoust. Soc. Am. Suppl. 1 81, S78 (1987)] and sets the four parameters by an iterative matching process. The hardware realization, by a circuit involving coupled oscillators, invites comparison with analogous neural structures.

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987 TUTTLE CENTER ROOM, 1:00 TO 2:50 P.M.

Session Z. Physical Acoustics IV: Recent Advances in Fiber Optic Sensors

David L. Gardner, Chairman

National Oceanic and Atmospheric Administration, Code N/CGx3, Rockville, Maryland 20852

Chairman's Introduction—1:00

Invited Papers

1:05

Z1. Acoustic sensor development at NRL. A. Dandridge (Code 6574, Naval Research Laboratory, Washington, DC 20375)

The development of fiber optic acoustic sensors at the Naval Research Laboratory will be reviewed. The talk will describe the fundamentals of the design and operation of fiber optic sensors. The major area of this talk will be the transduction mechanism of acoustic energy to optical phase shift. Methods to detect the phase shifts and to provide a linear voltage output will only be briefly discussed. The frequency regimes to be covered range from a few tens of Hz to MHz. In the low-frequency regime (up to several kHz), where the pressure is hydrostatic, the role of the material moduli of fiber coatings and bulk mandrels will be considered. A number of sensor designs for operation in the higher-frequency regimes, where inertial effects do not permit direct axial strain terms to play a major role, will also be described.

1:35

Z2. Multiplexing techniques for interferometric fiber-optic sensors. A. D. Kersey and A. Dandridge (Code 6574, Naval Research Laboratory, Washington, DC 20375)

Considerable research interest is presently being directed toward the development of all-fiber, multisensor networks for use in arrays and applications where a large number of different physical parameters are of interest, such as those encountered in the industrial process control area. A number of "all-optical" approaches to the multiplexing of such networks have been proposed in recent years. These techniques are most attractive for use in a wide range of application areas, as the sensors operate passively and can be remotely located from the source and detection electronics. Furthermore, the overall system comprising the sensors and fiber links retains a low susceptibility to electromagnetic interference. The principle of operation of multiplexing techniques based on time, frequency, and coherence-division addressing schemes will be described. The paper will compare experimentally reported performance features such as the multiplexing efficiency, sensitivity, crosstalk, etc., and will discuss practical limitations of the various methods.

2:05

Z3. Thermal noise limitations in a fiber optic seismic sensor. S. L. Garrett, T. Hofer (Physics Department, Code 61 Gx, Naval Postgraduate School, Monterey, CA 93943), and D. L. Gardner (National Oceanic and Atmospheric Administration, Code N/CGx3, Rockville, MD 20852)

Optical fibers are neither sensitive nor selective as a transduction medium. The high sensitivity of most fiber optic sensors has been a consequence of the fact that optical interferometric demodulation is capable of resolving phase changes of order $1 \mu\text{rad}/\sqrt{\text{Hz}}$. In a fiber sensor of 10-m length, this is equivalent to a resolution of one part in 10^{14} during a 1-s observation using light of 900-nm vacuum wavelength. For comparison, a good quality 1-in. condenser microphone can resolve a relative change in diaphragm-to-backplate separation of one part in 10^8 during a 1-s observation. This detection advantage of 120 dB has obscured many "sins" in several fiber optic sensor designs. This presentation will concentrate on the sensitivity limitations imposed by the fluctuation-

dissipation theorem [L. D. Landau and E. M. Lifshitz, *Statistical Physics* (Addison-Wesley, Reading, MA, 1969), 2nd ed., Chap XII] at nonzero temperatures and will demonstrate that the thermal limitations are entirely analogous to fluctuations observed in "mirrored galvanometers" that also use "optical leverage" to increase sensitivity. The results will be applied to the performance of a seismic sensor with a sensitivity that is limited by its intrinsic thermal noise. [Work supported by the Office of Naval Technology and the Office of Naval Research.]

Contributed Paper

2:35

Z4. A compact fiber optic laser probe to monitor surface vibration displacements. Jacek Jarzynski, Dowon Lee, and Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

An heterodyne laser interferometer is used to monitor the phase modulation of the laser light scattered from a vibrating surface. A laser beam is split into two arms of an interferometer where light is guided through single mode optical fibers so as to preserve phase information. The refer-

ence and frequency-shifted signals are recombined in a multimode fiber tightly connected to a receiving photomultiplier. Miniature gradient refractive index (GRIN) lenses are used to provide maximum efficiency and to make the probe as compact as possible. Both normal and tangential components of the surface displacement can be determined with a probe illuminating the vibrating surface at normal incidence and at two mutually perpendicular off-normal directions. The tangential component of the surface displacement is found from speckle reflection of the nonperfectly flat vibrating surface. This compact probe can be positioned at some distance (10–50 cm) from the vibrating surface for truly nonintrusive measurements.

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987

PREFUNCTION AREA, 1:00 TO 3:00 P.M.

Session AA. Psychological and Physiological Acoustics IV (Poster Session)

Craig Formby, Chairman

Department of Communicative Disorders/Neurology, J. H. Miller Health Center, University of Florida, Box J-174, Gainesville, Florida 32610

Contributed Papers

All posters will be displayed from 1:00 to 3:00 p.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:00 to 3:00 p.m.

AA1. Discriminability, loudness, and masking in the rat: A confirmation and extension. Thomas G. Raslear (Department of Medical Neurosciences, Walter Reed Army Institute of Research, Washington, DC 20307-5100)

Rats discriminated between two sound-pressure levels (SPL) of a pure tone: standard (STD) SPLs of 84 and 74 dB, and comparison (CO) SPLs 4, 14, and 24 dB below STD were tested in quiet and 60-dB noise at 4 and 12.5 kHz (24 conditions). The decibel difference between STD and CO accounted for only 43.52% of the variance in the signal detection measure of sensitivity (d') across conditions, whereas the loudness difference ($LD = STD^{0.35} - CO^{0.35}$) accounted for 89.82% of the variance in d' . These results confirm and extend previous observations that: (1) equal decibel differences are not equally discriminable [R. Pierrel, J. G. Sherman, S. Blue, and F. Hegge, *J. Exp. Anal. Behav.* **13**, 17–35 (1970)]; (2) loudness for the rat increases as a power function of SPL, with an exponent of 0.35 [R. Pierrel-Sorrentino and T. G. Raslear, *J. Comp. Physiol. Psychol.* **94**, 757–766 (1980)]; (3) masked loudness is a linear function of loudness in quiet [T. G. Raslear, R. Pierrel-Sorrentino, and F. Rudnick, *Behav. Neurosci.* **97**, 392–398 (1983)].

AA2. Evidence for an acoustical pathway to the inner ear through the lower jaw for an echolocating dolphin (*Tursiops truncatus*). Randall

L. Brill (Chicago Zoological Society, Brookfield Zoo, Brookfield, IL 60513 and Parmlly Hearing Institute, Loyola University of Chicago, 6525 N. Sheridan Road, Chicago, IL 60626), Martha L. Sevenich, Timothy J. Sullivan, Janet D. Sustman, and Ronald E. Witt (Chicago Zoological Society, Brookfield Zoo, Brookfield, IL 60513)

A dolphin (*Tursiops truncatus*) was conditioned to perform a discrimination task, by means of echolocation, in a "Go/No-go" paradigm while stationed in an underwater hoop and wearing suction cups over its eyes. To investigate the hypothesis that the fat-filled lower jaw of odontocete cetaceans provides an acoustical pathway to the inner ear, the dolphin was additionally required to perform the task while wearing either of two rubber hoods designed to cover its lower jaw. One hood, constructed from nonfoamed neoprene, allowed returning acoustic signals to pass, while the other hood, constructed from closed-cell neoprene, substantially attenuated such signals. The dolphin's performance was significantly hindered while wearing the attenuating hood ($p < 0.001$, X^2) as opposed to its wearing the hood made of nonfoamed neoprene or no hood at all. Acoustical data tape recorded during the experiment indicate that, with few exceptions, the spectral peaks of the dolphin's outgoing echolocation signals averaged between 30 and 50 kHz. These results support the hypothesis and agree with previous electrophysiological data that suggest that the lower jaw is involved in the transmission of high-frequency signals to the inner ear.

AA3. Modulation and gap detection with filtered and broadband noise signals. C. Formby (Departments of Communicative Disorders and Neurology, University of Florida, Gainesville, FL 32610), K. Muir, J. Uranga, C. Lynn, and A. Ahlers (Department of Speech, University of Florida, Gainesville, FL 32610)

The higher audiometric frequencies arguably are the most important for some tasks requiring fine temporal discriminations. Both modulation and gap detection thresholds improve with increasing center frequency of noise bands. For modulation detection, however, not all investigators are convinced that improved performance is necessarily dependent on sensitivity at the higher center frequencies. They point out that the high-frequency advantage for modulation detection may be a consequence of concomitant increases in stimulus bandwidth as a function of increasing center frequency of the noise band. To determine whether signal frequency or signal bandwidth is more important for modulation and gap detection, modulation and gap detection thresholds were measured for low-pass (LP) filtered noise ($f_c = 4000$ Hz, bandwidth = 4000 Hz), high-pass (HP) filtered noise ($f_c = 4000$ Hz, bandwidth = 2500 Hz), and broadband (BB) noise (bandwidth = 6500 Hz). Modulation and gap detection thresholds for the BB and HP signals were similar, and were better than thresholds for the LP signal. The ability to hear the higher frequencies of the noise signal apparently is more important for these tasks than is perception of signal bandwidth. [Research supported by Division of Sponsored Research, University of Florida, and American Cancer Society.]

AA4. Effect of click rate and delay on breakdown of the precedence effect. Rachel K. Clifton (Department of Psychology, University of Massachusetts, Amherst, MA 01003) and Richard L. Freyman (Department of Communication Disorders, University of Massachusetts, Amherst, MA 01003)

When clicks are presented from two loudspeakers, with one output leading the other by a few ms, a listener localizes the sound solely at the leading loudspeaker, a phenomenon known as the precedence effect. However, the precedence effect can be disrupted for several seconds if the leading and lagging (echo) locations are quickly switched [Clifton, unpublished manuscript]. Click trains were presented from two loudspeakers, placed 90° right and left off midline in an anechoic chamber, with lagging click delays ranging from 2–9 ms. Click rates were 1, 2, 4, 6, and 8/s. Under some conditions, subjects reported hearing clicks from both loudspeakers immediately following the switch in leading and lagging outputs, indicating a breakdown of the precedence effect. The length of time both clicks were heard following the switch was affected by both echo delay and click rate. Inhibition of the precedence effect was strongest when rate of clicks was slower and delays were longer, that is, just under the subject's echo threshold. At shorter delays and faster rates the precedence effect was experienced throughout the trial.

AA5. Inhibitory mechanisms and cortical processing. D. M. Daly, D. D. Daly (Box 210855, Dallas, TX 75211), J. W. Drane (Auburn University, Auburn, AL 36849), and J. A. Wada (University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada)

When balance of inhibitions declines/fails, patients with seizures in auditory cortex report certain synthetic [be]–[de]–[ge] as nasals, “bleats,” and then undifferentiated buzzes. Effects of increased inhibition can appear with auditory cortex in volume surrounding disinhibited areas (even contralaterally): Patients report same range of stimuli as clearly identical [J. Neurophysiol. 44, 200–222 (1980)]. Here a case with seizures in visual cortex and post-ictally impaired auditory comprehension (but normal peripheral acuity), is reported. Testing included prerecorded sets of [be]–[de]–[ge], [ge]–[ye], and [be]–[we] presented through headphones monaurally and binaurally. During episodes with more frequent seizures, patient had intervals (some lasting nearly an hour) when [de] boundaries for [be]–[de] and [de]–[ge] increased approximately 200 Hz; [be]–[we] and [ge]–[ye] boundaries increased 40–50 ms (with stimuli less than 50 ms now reported as vocalic). The BDG stimuli cover tonotopically differing extents of cortex; BW or GY stimuli cover tonoto-

pically limited extents at different rates. Results appear consistent with augmenting infield inhibition.

AA6. The effects of pedestal level on incremental auditory responses in humans: Loudness matching study. Hiroshi Riquimaroux and Peter Dallos (Auditory Physiology Laboratory, Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

The effects of pedestal level (L_B) on incremental auditory responses were investigated in normal-hearing human subjects with amplitude modulated (AM) white noise by means of a loudness matching procedure. The method of adjustment was used to find equally loud increments (ΔL) for various standards ΔL s and L_B s (30–80 dB SPL). The L_B for standard ΔL was kept constant at 30 dB SPL. The hypothesis to be tested was that loudness of ΔL is independent of L_B . The results show that loudness of ΔL appears to be almost independent of L_B for very small ΔL (<6 dB). The findings also indicate the possibility that the stimulus peak level (L_P) might be involved in the determination of loudness of ΔL when ΔL is large. The data demonstrate that the slope of loudness growth function increases as L_B increases. For low L_B s, the slope has a tendency to asymptote to the reference line with the slope of one. The results agree with earlier JND data. It is concluded that the hypothesis appears to be supported by the present data for very small ΔL s.

AA7. The effects of pedestal level on incremental auditory responses in humans: Brainstem response (ABR) study. Hiroshi Riquimaroux and Peter Dallos (Auditory Physiology Laboratory, Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

The effects of pedestal level (L_B) on incremental auditory responses were examined in normal hearing human adults with amplitude modulated (AM) white noise by using the auditory brainstem response (ABR). The wave V latency of ABR, yielded by a noise increment (ΔL) with various L_B s (30–80 dB SPL), was measured. The hypothesis of this study is that the wave V latency derived by ΔL is invariant with L_B . The results show that the wave V latency appears to be almost independent of L_B for very small ΔL (<6 dB). The data are compared with those of the companion loudness matching study. Similar tendencies are seen in the characteristics of iso-wave V latency contours and equal loudness contours for relatively small ΔL s or for low L_B s. The growth functions obtained from the two experiments, however, are different. In the loudness matching study the slope of loudness growth functions increases as L_B increases, while in the ABR study the slope of growth functions of wave V latency increases only for low L_B .

AA8. Testing children and the aged with computerized audiometry. Michel Picard (University of Montreal, Montreal, Quebec H3C 3J7, Canada), Henry J. Ilecki, and James D. Baxter (Royal Victoria Hospital/McGill University, Montreal, Quebec H3A 1A1, Canada)

Using computerized audiometry in a group of noise-exposed workers has been shown to be a reliable and valid means of assessing hearing sensitivity [M. Picard *et al.*, *Proceedings: Computer Applications to Canadian Health*, pp. 127–130 (1987)]. Based on this study, an investigation was conducted to similarly determine the feasibility of implementing like-methodology in two clinical populations typically regarded as “difficult to test” (*viz.*, geriatric and pediatric groups). Children were aged 8 to 12; seniors, 65–81 years. Reliability was assessed by relating air- and bone-conduction thresholds where there was no evidence of middle-ear pathology. Validity was measured by comparing air-conduction pure-tone thresholds (0.5, 1, 2, 3, 4, and 6 kHz) obtained under two test modalities, computerized audiometry and conventional testing performed by a trained examiner. Both procedures followed ANSI S3.21-1978. Findings in both age groups were comparable to those reported in the previous study. Coefficients of reliability remained equally high across frequencies regardless of testing method. As well, high correlations between conventional and computerized audiometry were found. Complementing this

observation, a PCA supported the criterion-related validity of computerized audiometry. [Work supported by MRC.]

AA9. Frequency specificity of the cochlear action potential varies with stimulus presentation rate. Alex N. Salt, Arti R. Vora, and Ruediger Thalmann (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

It is well documented that the amplitude of an averaged cochlear action potential decreases as the rate of stimulus presentation is increased. In the present study, frequency regions contributing to the AP elicited by tone burst stimuli at rates from 4 to 80/s have been investigated in guinea pigs using a tone masking procedure. Derived responses corresponding to specific regions were obtained by subtracting AP responses in the presence of a low-level continuous masking tone from those in the absence of masker. This is a modification of the technique described by Hood *et al.* [J. Acoust. Soc. Am. Suppl. 1 **81**, S8 (1987)]. When stimulation rate is increased, the derived response amplitudes decreased near the probe frequency region, while amplitudes at outlying frequencies were less markedly changed. These data suggest the contribution to the AP from the region corresponding to the probe frequency becomes less at high stimulation rates. With an 80/s stimulation rate the contribution from this region is negligible. The degradation of frequency specificity of responses is presumed to arise from each stimulus acting as a "forward masker" for the following stimulus. To optimize the frequency specificity of AP responses to tone burst stimuli, the rate of presentation must therefore be considered. [Work supported by NIH.]

AA10. Proteins of the organ of Corti (OC). Isolde Thalmann, Gertraud Thallinger, and Ruediger Thalmann (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

The majority of proteins found to date in OC were identified by immunohistological methods. While these techniques are powerful tools for localization of proteins, they frequently lack satisfactory specificity. Two-dimensional polyacrylamide gel electrophoresis (2D-PAGE) is able to

demonstrate the presence in the OC of specific subtypes of certain proteins. In the case of contraction associated proteins, for instance, the presence of nonmuscle actin (beta and gamma), three types of nonmuscle tropomyosin (Tm:4, Tm:5, and Tm:8) and nonmuscle lactic dehydrogenase (LDH B) was documented. Another important role of 2D-PAGE is the demonstration of previously unknown proteins. The presence in the OC of two highly prominent proteins of low molecular weight and strongly acidic pH (assumed to be specific to the OC) was shown. Accordingly, they were termed OCP-I and OCP-II [I. Thalmann *et al.*, Arch. Otorhinolaryngol. **226**, 123-128 (1980)]. It was found that small amounts of OCP-I and OCP-II are present in spiral ligament and spiral limbus and more substantial amounts in basilar membrane. Neither protein is detectable in stria vascularis. Likewise, no proteins resembling OCP-I or OCP-II were found in tissues other than inner ear. While some biochemical features of the two proteins have in the meantime been characterized, their functional significance remains obscure. [Work supported by NIH.]

AA11. Ultrastructural correlates of hair cell motility. B. N. Evans and A. Yonovitz (Graduate School of Biomedical Sciences, The University of Texas Health Science Center at Houston, Houston, TX 77030)

Nonstimulated as well as outer hair cells showing mechanical displacements to electrical and chemical stimulation were examined ultrastructurally to assess the lability of a rhodamine-labeled actin structure extending between the cuticular plate and the peri-nuclear region. Hair cells of varying lengths (20-80 μm) were stimulated and immediately fixed following mechanical displacement. All cells revealed a distinct non-membrane bound complex in cross section, but were infrequently observed in longitudinal sections. The cross-sectional area of the structure increased in successive ascending sections. This structure spirals apically in a nondirect manner, but eventually consolidates into a primary mass to become contiguous with the inner leaflet of the cells plasmalemma. A localized microtubule network was found to pass most prominently in the thin basal aspect of the complex. The cross-sectional orientation appears to optimize the visualization of this potentially important aspect of the outer hair cells anatomy.

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987 UM LEARNING CENTER, 1:00 TO 2:15 P.M.

Session BB. Psychological and Physiological Acoustics V: Frequency Discrimination and Pitch Perception

William M. Hartmann, Chairman

Physics Department, Michigan State University, East Lansing, Michigan 48824

Contributed Papers

1:00

BB1. Tests of a modified energy-detector model of frequency discrimination. David S. Emmerich, Wolfgang Ellermeier, and Brenda Butensky (Department of Psychology, State University of New York at Stony Brook, Stony Brook, NY 11794)

Henning [J. Acoust. Soc. Am. **42**, 1325-1334 (1967)] proposed a modified energy-detector model of auditory discrimination and detection. This model, as well as several other models, predicts that random variations of the amplitudes of the signals to be discriminated in a frequency-discrimination experiment should make the task much harder. However,

Henning [J. Acoust. Soc. Am. **39**, 336-339 (1966)] reported that, for signal frequencies less than 4000 Hz, variations in amplitude did not lead to the expected impairment on the frequency-discrimination task. A reexamination of this question indicates that when the appropriate control comparisons are made, subjects do in fact show the predicted impairment. Nevertheless a second experiment suggests that this result may not provide a good test of Henning's (1967) energy detection model since variations in pitch with intensity, which are not considered in the model, are likely to be involved. For this reason, a third experiment was conducted in order to provide an independent test of the model. The results of this third experiment do not support the energy-detector model.

1:15

BB2. Detecting dynamic frequency shifts of simple and complex tones. Robert P. Carlyon and Richard J. Stubbs (MRC Institute of Hearing Research, University Park, Nottingham NG7 2RD, United Kingdom)

Thresholds for the detection of frequency modulation by a single cycle of a 12-Hz sinusoid were obtained for carriers that were simple or complex tones. Two phases of modulator were used, producing percepts of either upward or downward frequency glides during the middle of the stimulus. To ensure that subjects based detection on frequency *movement*, both the phase of the modulating sinusoid and the starting frequencies were randomized from presentation to presentation. For pure-tone carriers, thresholds (= minimum detectable frequency swing/carrier frequency) were constant for carrier frequencies of 800–2400 Hz and increased below 800 Hz. Thresholds were lower for modulations producing the percept of an upward glide than for “downward” glides. This difference was much greater at low carrier frequencies. When the carrier was a three-component harmonic complex, thresholds were almost independent of its center frequency but increased with decreasing fundamental frequency. Thresholds for harmonic and for inharmonic complexes were compared. These results have implications for the recording of voice pitch (e.g., in signal-processing hearing aids).

1:30

BB3. Central spectrum models of dichotic edge pitches. W. M. Hartmann (Physics Department, Michigan State University, East Lansing, MI 48824)

The central spectrum model of Raatgever and Bilsen [J. Acoust. Soc. Am. 80, 429–441 (1986)] successfully accounts for a number of dichotic-noise pitch effects and for the lateralization of the pitches. The model is, however, physiologically implausible. An improved model, based upon a neural delay line having coincidence cells that obey the statistical rules of Stern and Colburn [J. Acoust. Soc. Am. 64, 127–140 (1978)], retains the predictive power of the Raatgever–Bilsen model. It also successfully predicts the results of new experiments where the Raatgever–Bilsen model fails. New experiments measure the relative strengths of Huggins pitches $HP+$ and $HP-$, the lateralization of the binaural-edge pitch (BEP), the spectral density dependence of the BEP, and the strength of generalized BEP, where the interaural phase shift may have any magnitude. [Work supported by the NIH.]

1:45

BB4. Infants' extraction of pitch from inharmonic tonal complexes. Marsha G. Clarkson and Rachel K. Clifton (Psychology Department, University of Massachusetts, Amherst, MA 01002)

Previous research has demonstrated the infant's ability to extract the virtual pitch of harmonic tonal complexes. In the present study, we used a conditioning procedure to assess 7-month-old infants' perception of pitch for inharmonic tonal complexes. Infants initially discriminated a pitch change from 160 to 200 Hz or vice versa in harmonic tonal complexes, which contained eight consecutive harmonics but no fundamental frequency. After infants discriminated the harmonic complexes, each partial's frequency was shifted upward by 30 Hz to render the sounds inharmonic. All infants who discriminated the inharmonic complexes proceeded to a categorization stage in which a randomized sequence of three inharmonic complexes (containing six partials) comprised each pitch category so that the spectral characteristics of the tokens changed continuously while the pitch category changed only on change trials. The harmonic complexes were discriminated by 19 of 22 infants, while 16 infants successfully discriminated the inharmonic sounds. However, only 11 infants categorized spectrally different sounds consistently with their pitches, suggesting that infants had more difficulty extracting pitch from inharmonic complexes than from harmonic ones presented in previous research. [Work supported by NSF and NICHD.]

2:00

BB5. Pitch and timbre of inharmonic tone complexes of Balinese “Gender Wayang” instruments. Werner A. Deutsch (Department of Sound, Austrian Academy of Science, A-1010, Vienna, Austria) and Franz Födermayr (Department of Musicology, University of Vienna, A-1010 Vienna, Austria)

The tones of the scale of two Balinese “Gender Wayang” instruments have been stored on computer disk and are used as test tones in pitch detection experiments. The subjects were instructed to identify the pitch of the inharmonic instrument tones by tuning a sinusoidal test tone on the appropriate pitch height in a computer controlled up/down adaptive procedure. The experimental results support the general ambiguity of pitch perception in case of inharmonic tone complexes, though a certain preference of the subjects on the lowest partial could be observed. This perceptual ambiguity could be increased further when the first partial of the instrument tones was removed by means of narrow-band digital filtering. Based on frequency analysis data of the natural sounds, a synthesis of the figuration “melody” of the Gender Wayang piece “Lagudelem” has been performed in which the inharmonic complex tones have been replaced by harmonic ones. The comparison of all three, the original sound probes and the filtered and the synthesized tones shows the strong dependence of pitch on timbre. The results of the pitch detection experiments are discussed on the basis of critical band analysis and the current theory of pitch perception as published by E. Terhardt (1972).

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987

UMS 4, 1:00 TO 3:05 P.M.

Session CC. Underwater Acoustics IV: Wedge Model Propagation

Ding Lee, Chairman

Naval Underwater Systems Center, New London, Connecticut 06320

Chairman's Introduction—1:00

Contributed Papers

1:05

CC1. Range-dependent cut-on frequencies in a three-dimensional ocean environment. Stewart A. L. Glegg and Jong R. Yoon (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

Sound propagation in a three-dimensional wedge-shaped ocean envi-

ronment demonstrates significant ray curvature in a direction parallel to the wedge apex. The theoretical prediction of this shows that these effects are more significant at low frequencies than at high frequencies; consequently, the three-dimensional ocean acts as a range-dependent acoustic filter. This paper describes an experimental and theoretical investigation of pulse propagation in a model scale three-dimensional wedge-shaped

ocean. In the experimental setup the complete acoustic field could be measured for a given source position and the wedge angle could be changed. The results show good agreement with the idealized modal solution of Buckingham, and demonstrate how the pulse is distorted as a function of range. They also demonstrate how the cut-on frequency for acoustic propagation increases as a function of range parallel to the shoreline.

1:20

CC2. Scattering in basins or horizontally refracting modes: Which is it? C. H. Harrison (CAP Scientific, 40-44 Coombe Road, New Malden, Surrey KT3 4QF, United Kingdom)

In a large basin there are at least two types of 3-D ray paths from a source to a distant receiver via the sloping basin edge (other than the direct path). One is the simple scattered ray at the seabed, and the other is the multiple reflected ray that steepens in shallow water but finally turns in the horizontal plane towards deep water (otherwise interpreted as a horizontally refracting vertical mode contribution [C. H. Harrison, *J. Acoust. Soc. Am.* **65**, 56-61 (1979)]). If the total change in heading of the latter ray is 2θ , its elevation angle will never exceed θ [C. H. Harrison, *J. Acoust. Soc. Am.* **62**, 1382-1388 (1977)], and the path may exist despite reflection losses in realistic environments. Therefore, if the top and bottom surfaces are perfectly smooth, there will be no scattering and there will certainly be multiple reflection. Conversely, if the surfaces are rough, the multiple reflection will be inhibited but there will definitely be a scattered return. Since there is scope for misinterpretation of experimental results, this paper proposes some experiments to distinguish the two paths.

1:35

CC3. Diffraction of sound by hard strips and truncated wedges. Ivan Tolstoy (Knockvennie, Castle Douglas, SW Scotland DG7 3PA)

Starting from rigorous diffraction solutions for single, hard, semi-infinite wedges, it has long been possible to construct approximate solutions for more complex angular surfaces [A. D. Pierce, *J. Acoust. Soc. Am.* **55**, 941-955 (1974)]. Demonstrated here is an exact procedure that formulates the multiple scatter between vertices of an angular body via the standard self-consistent algorithm. This allows one to obtain the correct diffracted field of each vertex. Summing these together with the relevant incident and image fields leads to an exact solution for the full sound field. This is done here for the case of a plane harmonic incident sound field for (1) a truncated wedge and (2) a hard strip of width l . Numerical results for the latter model are shown to compare satisfactorily for all kl with experimental data [H. Medwin *et al.*, *J. Acoust. Soc. Am.* **72**, 1005-1013 (1982)]. This theoretical procedure yields, in the plane wave case, a formal estimate of the error incurred in the "double diffraction" approximation (i.e., the procedure that neglects all but the first two terms of the multiple scatter series). This error decreases like e^{kl}/kl for increasing kl . It is then possible to confirm theoretically the observation of Medwin *et al.* who showed that, for the geometry of their strip experiment, this approximation is actually very good for $kl \gtrsim 3$ and quite adequate in the region $1 < kl < 3$, using the Biot-Tolstoy rigorous impulsive point source solution for the perfect wedge [M. A. Biot and I. Tolstoy, *J. Acoust. Soc. Am.* **29**, 381-391 (1957)] for their double-diffraction solution. [Work supported by ONR.]

1:50

CC4. An accurate numerical solution to ASA Benchmark Problem I using the finite element method. Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148) and Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529-5004)

A Special Session on Numerical Solutions of Two Benchmark Problems was presented at the 113th Meeting of the Acoustical Society of

America in Indianapolis, Indiana, 11-15 May 1987. The underwater acoustics community was invited to present solutions to these problems and compare them against a numerically "exact" solution obtained from the R. B. Evans' coupled mode model [F. B. Jensen, *J. Acoust. Soc. Am. Suppl.* **1** **81**, S40 (1987)]. Our finite element ocean acoustic propagation model [J. E. Murphy and S. A. Chin-Bing, *J. Acoust. Soc. Am. Suppl.* **1** **81**, S9 (1987)] to Problem I is applied here. (Problem I dealt with shallow water upslope propagation in a wedge-shaped channel.) This finite element model gives a full-wave solution for range-dependent acoustic propagation with accuracy that rivals coupled mode models. The model's capability will be demonstrated by presenting a numerical solution to Problem I that compares favorably to the Evans' coupled mode model solution. [Work supported by ONR/NORDA.]

2:05

CC5. Sound interaction with a sloping, upward-refracting ocean bottom. Hassan B. Ali, Juan I. Arvelo (Naval Surface Weapons Center, Code U-25, White Oak, Silver Spring, MD 20903-5000), Anton Nagl, and Herbert Überall (Department of Physics, Catholic University, Washington, DC 20064)

Results of a parabolic-equation (PE) code are shown for the problem of sound interaction with an upsloping, upward-refracting ocean bottom. They are compared with the results of a coupled-mode code [J. F. Miller *et al.*, *J. Acoust. Soc. Am.* **79**, 562 (1968)] for a range-dependent environment. This includes bottom penetration effects at mode cutoff and a study of how these are affected by the sound-speed gradient and by the absorption in the sediment. The bottom penetration features are further illuminated by employing Gaussian sound pulses and studying their propagation, illustrating the effects of bottom penetration and reradiation back up into the water (as caused by the upward-refracting bottom gradient), arriving ahead of the water-borne pulse.

2:20

CC6. Spherically symmetric acoustic propagation across a fluid/fluid boundary. Michael J. Buckingham (Department of Ocean Engineering, Room 5-204, Massachusetts Institute of Technology, Cambridge, MA 02139)

The acoustic field produced by a point source at the center of a fluid sphere, which itself is immersed in a fluid medium, is of interest in connection with ambient noise in the Arctic Ocean. A solution for the field in the internal and external fluid domains is presented based on novel finite Hankel transforms. The method involves internal and external Hankel transformations of the Helmholtz equation, which introduce explicitly the boundary values of the field and its spatial derivative at the spherical interface between the two fluids. By combining the resultant equations with the boundary conditions (continuity of pressure and normal component of velocity) all the unknown constants are determined, and on performing the inverse Hankel transforms, the final expression for the field is obtained. The mathematical formalism of the finite Hankel transforms introduced here has the advantage of being able to handle complicated boundary conditions, whereas previous finite Hankel transform techniques are limited to problems involving Dirichlet, Neumann, or mixed (i.e., impedance) boundaries. [This work was supported by the Office of Naval Research, Contract No. N00014-86-K-0325.]

2:35

CC7. Sound transmission experiments from an impulsive source near rigid wedges. Saimu Li (Shandong College of Oceanography, Quindao, Shandong, People's Republic of China) and C. S. Clay (Geophysical and Polar Research Center, University of Wisconsin, Madison, WI 53706)

Experimental sound transmissions in air from a spark source to a small microphone were made near rigid wedges. Two types of experiments were made. The first experiments were transmissions within an approximately 12° wedge and a 52° wedge. The Biot-Tolstoy wedge solution [I. Tolstoy, *Wave Propagation* (McGraw-Hill, New York, 1973)] was used to calculate the theoretical impulse response. The "free air"

transmission from the spark source was convolved with the theoretical transmission. The transmissions within the wedge gave finite sets of multiple reflections or image arrivals and a diffraction arrival. The diffraction was sensitive to leaks at the wedge apex. Theory and experiment matched. The second set of experiments was made near a 270° wedge. Arrivals were the direct reflection and the diffraction. The image reflection when the specular "reflection point" was very near the wedge apex was of interest. Comparisons of data and theoretical signals using Biot-Tolstoy theory were excellent. Theoretical diffraction signals calculated with Trorey's theory had poor matches [A. W. Trorey, *Geophysics* 35, 762-784 (1970)].

2:50

CC8. Impulse response of density contrast wedge using normal coordinates. D. Chu and C. S. Clay (Geophysical and Polar Research Center, University of Wisconsin, Madison, WI 53706)

The exact impulse response of a point source to a rigid wedge was derived by Biot and Tolstoy [I. Tolstoy, *Wave Propagation* (McGraw-Hill, New York, 1973)]. The solution by the experiments has been verified. The wedge having free (pressure release) boundaries has the same form as the rigid boundary solution [W. A. Kinney *et al.*, *J. Acoust. Soc. Am.* 73, 183-194 (1983)]. These solutions cannot be applied directly to many actual cases such as ocean floor, where the ratio of density of the base sediment to that of the water approaches neither infinite (rigid) nor zero (pressure release). The solution is the impulse response to an isovelocity ($v_1 = v_2$) and density contrast ($\rho_1 \neq \rho_2$) wedge. It is an extension of the solution given by Biot and Tolstoy. The reflection part of the solution is an impulse series weighted by reflection coefficient with different order. The diffraction part is the summation of the diffractions due to the individual image sources. The amplitudes are a function of the reflection coefficient and the number of multiple reflections. [This work was partly supported by the People's Republic of China and the Office of Naval Research.]

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987

UMS 5, 1:00 TO 3:05 P.M.

Session DD. Underwater Acoustics V: Noise Modeling

Tsih C. Yang, Chairman

Naval Research Laboratory, Code 5123, Washington, DC 20375

Chairman's Introduction—1:00

Contributed Papers

1:05

DD1. Low-frequency noise and bubble plume oscillations. W. M. Carey (Naval Underwater Systems Center, New London, CT 06320) and J. W. Fitzgerald (Consultant, New London, CT 06320)

The evidence for a wind dependence in low-frequency ambient noise (< 200 Hz) is sparse due to the corrupting influences of the radiated sound from ships. Measurements made with hydrophones below critical depth, in sparse shipped basins, or at high sea states show indications of two distinct regimes associated with the occurrence of breaking waves and a variation of root-mean-square pressure with the square of the local wind speed. In a previous paper [W. Carey, *J. Acoust. Soc. Am. Suppl.* 1 78, S1 (1985)], the generation of sound by wave turbulence interaction at low sea states (WS < 10 m/s) and collective bubble oscillations driven by turbulence at high sea states was proposed (WS > 10 m/s). This paper shows that recently observed bubble clouds that penetrate to tens of meters below the surface of the sea result in regions of low sonic velocity described by Wood's treatment of air-bubble water mixtures, but with a density close to that of water. It is shown that such a region can be treated as a flexible body with mixture speed and density. The radiation from such a body would be monopole and dipole in nature; but due to the proximity of the sea surface, only the resulting dipole source would be of importance. These regions could also have a resonant characteristic, and when driven by the turbulence result in sufficient radiated sound. These results are similar to and consistent with the analysis by A. Prosperetti [Nato ASI, "Natural Mechanisms of Surface Generated Noise in the Ocean" (1987)].

1:20

DD2. Effect of longitude on vertical distribution of ambient noise due to wind and coastal shipping. F. H. Fisher and W. S. Hodgkiss (Marine Physical Laboratory, Scripps Institution of Oceanography, University of California—San Diego, La Jolla, CA 92093)

Using multielement vertical arrays deployed from FLIP, measurements of the vertical distribution of ambient noise have been made at a latitude of 32N at three stations, 124W, 136W, and 150W, the latter station being about 1700 miles from the coast. The arrays used were deployed

at the sound channel axis and were uniformly spaced at half-wavelength either at 200 or 300 Hz. From 75 to 300 Hz, the change in the vertical distribution of noise shifts, especially at higher frequencies, with increasing range from coastal shipping was in a manner consistent with the effect of chemical absorption on low-angle noise due to coastal shipping. Whereas a shipping noise pedestal at low angles is observed at all frequencies at short range, at long range the absorption effect makes the vertical distribution of ambient noise more isotropic at higher frequencies than that at low frequencies. Effects of wind speed on the vertical distribution of noise were also measured. As wind speed increases, ambient noise at angles greater than 15° from the horizontal (that is, noise of local origin) increases and approaches the levels of the pedestal at 124° W and presumably exceeds them at high wind speeds. Our results are orthogonal to those obtained by Kibblewhite *et al.*, ARL/UT, with their deployment at 150 W of three Vedabs bouys on a north-south line to measure low-frequency absorption using northern shipping as a sound source. This conflict suggests a simultaneous north-south and east-west experiment as well as the utility of using a multielement vertical DIFAR array. [Work supported by ONR.]

1:35

DD3. Estimation of surface noise source level from low-frequency seismoacoustic ambient noise measurements. Henrik Schmidt (Massachusetts Institute of Technology, Department of Ocean Engineering 5-204, Cambridge, MA 02139) and W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375)

The extraction of noise source levels from ambient noise measurements requires accounting for the seismoacoustic propagation from the surface generated noise sources to the field measurement positions. At frequencies below 100 Hz, the waveguide nature of the environment strongly influences the distribution of ambient noise. Only when these propagation effects are considered can measurements from different experiments taken in different environments be combined to ascertain the global properties of surface noise sources. Here, such a previous analysis of a shallow water experiment [Schmidt *et al.*, in *Natural Mechanisms of Surface Generated Noise*, edited by B. Kerman (Reidel, Dordrecht, The Netherlands, 1988)] is used and the results are combined with Kibble-

white and Ewans' lower-frequency results [J. Acoust. Soc. Am. **78**, 981–994 (1985)], the latter also containing a summary of previous experimental results obtained by others. These data are treated in the same way by accounting for the environments using a wave theory model of distributed noise [W. A. Kuperman and F. Ingenito, J. Acoust. Soc. Am. **67**, 1988–1996 (1980)]. When the particular environments are used, the spread in the reprocessed noise levels is substantially reduced, showing more consistency in the frequency dependence of the noise sources, suggesting that there are only a few (or possibly only one) dominant natural noise source mechanisms that primarily contribute to ocean ambient noise below 100 Hz. The importance of appropriately including propagation factors in processing noise data for estimating source levels is therefore strongly demonstrated.

1:50

DD4. Downslope propagation and vertical directionality of wind noise. W. M. Carey (Naval Underwater Systems Center, New London, CT 06320), R. E. Evans (ODSI, Stonington, CT 06378), and J. A. Davis (PSI, New London, CT 06320)

Measurements of the vertical noise intensity versus angle [W. Carey and R. Wagstaff, J. Acoust. Soc. Am. **80**, 1523–1526 (1986)] show the low-frequency distribution ($\ll 200$ Hz) to be broadly peaked about the horizontal, whereas the higher-frequency (≈ 400 Hz) distribution is peaked at the SOFAR angles ($\approx \pm 15^\circ$) near the axis with a minima at the horizontal. This effect has been attributed to the noise from ships over the basin margins [R. Wagstaff, J. Acoust. Soc. Am. **69**, 1009–1014 (1981)]. However, since spectral variation along the horizontal is smooth and the effect is observed in sparsely shipped areas of the world, wind noise must be included to explain this effect. R. W. Bannister [J. Acoust. Soc. Am. **79**, 41–48 (1986)] has attributed this effect to high-latitude winds and the shallowing sound channel found in Southern Hemisphere waters. W. Carey [J. Acoust. Soc. Am. **79**, 49–59 (1986)] attributed the effect to downslope propagation. Calculations between 50 and 400 Hz of the mid-basin vertical directionality made with ASTRAL and PAREQ with a geoacoustic model were both found to show that the bottom behaves as a low-pass filter as the low-frequency energy at higher angles interacts with the bottom and is converted to low-angle energy in the deep sound channel while, at the higher frequencies and higher angle, energy is absorbed. This low-pass effect is consistent with experimental results. Propagation of low-grazing angle energy from the shelf is prominent in the computational results but not apparent in measured data, perhaps due to bathymetric roughness.

2:05

DD5. Underwater noise produced by rainfall. H. C. Pumphrey and L. A. Crum (National Center for Physical Acoustics, The University of Mississippi, Oxford, MS 38677)

When a liquid drop falling through the air strikes a flat horizontal surface, there are acoustic emissions associated with this impact in both the air and the liquid [G. J. Franz, J. Acoust. Soc. Am. **31**, 1080 (1959)]. Under certain conditions, a small air bubble can be entrained in the liquid by the impacting drop and stimulated into volume pulsations that radiate quite strongly. When several drops impact the surface in a short time interval as in rainfall, the acoustic emissions associated with the bubble oscillations appear to be a major contribution to the total acoustic emission over a considerable bandwidth. With artificially created rainfall, the spectral peak observed near 20 kHz for natural rainfall can be duplicated [Scrimger *et al.*, J. Acoust. Soc. Am. **81**, 79 (1987)]. However, with the addition of surface tension reduction agents, this peak is significantly reduced, implying that a major portion of the noise associated with rainfall is related to oscillations of the entrained bubble rather than due to the impact of the drop with the liquid surface. [Work supported by the ONR.]

2:20

DD6. Correlation and spectra of ambient sound generated by wind, whitecaps, and rain. David Shonting (Naval Underwater Systems, Center, Newport, RI 02841), Foster Middleton (University of Rhode Island, Kingston, RI 02881), and Nancy Taylor (Applied Science Associates, Narragansett, RI 02882)

Although high correlation exists between oceanic ambient sound and

wind speed and rain rate, the sound generation mechanisms remain obscure. This is due partly to the difficulty of simultaneously measuring the sound-generating phenomena at the location of the acoustic measurements. Also, there is a lack of frequency spectra of sound generated by the specific sources that may distinguish between wind- and rain (or spray)-generated sound and may hold important clues to the generating mechanisms. A pier facility was established on Narragansett Bay to monitor broadband sound associated with observed sea surface phenomena. Simultaneous recording of wind velocity, wave height, and rain rate allows precise comparison of the environmental variables with their acoustic response. In addition, visual observation is made of whitecapping and of interfering ship traffic. The ambient sound spectra are routinely generated as high-resolution FFT estimates on a MASSCOMP computer. Preliminary measurements show sound pressure to vary linearly with wind speed, with correlations ranging from 0.88–0.97. The slope changes around wind speeds of 6–7 m/s, i.e., where whitecaps form. The spectra of wind-generated sound display a faster roll-off than rain-produced sound, the latter exhibiting a broad peak around 15–18 kHz. Plots are presented of the intense insonification of the water column by rapidly advecting rain squalls. [Work supported by NAVSEA/AIRSEA systems commands.]

2:35

DD7. Observations of deep-ocean ambient noise using a seismometer array. Anthony E. Schreiner, LeRoy M. Dorman, John A. Hildebrand, Dalia Lahav (Scripps Institution of Oceanography, University of California–San Diego, La Jolla, CA 92037), and Dale Bibee (NORDA, NSTL Station, MS 39529)

An array of 12 ocean-bottom seismometers (OBS) was deployed this past March in the deep ocean near San Diego. The array had a maximum extent of 150 m and a minimum interelement spacing of 8 m. The accuracy of the relative sensor positions was as good as 2 m in some cases. The seismometers were deployed in 3.8 km of water with a partially maneuverable vehicle at the end of a wire guided by a transponder net. The OBS capsules contained a vertical and two horizontal seismometers and a hydrophone. Three of the instruments had a low-frequency hydrophone of the Cox design. Three-min windows were recorded at intervals of 3 or 6 h for up to 30 days. Spectral levels have been calculated between 0.01 and 30 Hz. A power level drop of up to 40 dB below the microseism peak at frequencies below 0.1 Hz was observed. Similar spectra have been reported by Webb and Cox [J. Geophys. Res. **91**, 7343–7358 (1986)]. At the time of the abstract deadline, correlation was being calculated as a function of sensor separation to determine the correlation length of the ambient noise.

2:50

DD8. Ambient ULF/VLF ocean-bottom noise 200 km west of San Francisco: Motion and pressure 0.002 to 20 Hz. G. H. Sutton, J. A. Carter, and N. Barstow (Rondout Associates, Incorporated, P. O. Box 224, Stone Ridge, NY 12484)

Data recorded from long-period and short-period, three-component seismometers and hydrophones of the Columbia-Point Arena ocean-bottom seismic station (OBSS) are being analyzed in the frequency range 0.001 to 40 Hz to determine the characteristics of ULF/VLF ocean-bottom noise. The OBSS, located at $38^\circ 09.2' N$, $124^\circ 54.4' W$ at 3903-m depth, operated for over 6 years, from May 1966 to September 1972. Selected portions of essentially continuous FM tape data are being digitized to obtain ULF/VLF spectra and covariances during quiet and noisy times and during passage of vessels and earthquake wavetrains. It is planned to compare OBSS data with available source information on sea/swell conditions, tidal currents (measured by a current meter on the OBSS), earthquakes, and shipping. Spectral levels appear to be above system noise for most frequencies above about 0.002 Hz. The pressure spectra agree with those obtained more recently [Cox *et al.*, Atmos. Oceanic Tech. **1**, 237–246 (1984)], showing a strong minimum between about 0.03 and 0.1 Hz. The motion spectra from the OBSS seismometers are similar. Coherency between vertical motion and pressure is well above random for most of the spectrum below 0.5 Hz. There is clear evidence that ambient noise maxima rise and fall with weather conditions and water wave activity and that ULF noise, especially on the horizontal seismometers, has a tidal dependence. [Research supported by ONR.]

Session EE. Noise IV and Engineering Acoustics III: Application of Math Models and Experimental Investigations to Solve Acoustic Problems (Poster Session)

Larry H. Royster, Chairman

1217 BR Hall, MAE Department, North Carolina State University, Raleigh, North Carolina 27695-7910

Contributed Papers

All posters will be displayed from 1:30 to 3:00 p.m. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 to 2:15 p.m. and contributors of even-numbered papers will be at their posters from 2:15 to 3:00 p.m.

EE1. Boundary element solution for a coupled elastodynamic and wave equation system to predict forced response of a plugged acoustic cavity. Robert Ciskowski (International Business Machines, Rochester, MN 55901) and Larry H. Royster (Department of Mechanical Engineering, North Carolina State University, Raleigh, NC 27695-7910)

The subject of this investigation is the use of the boundary element method (BEM) to predict the steady-state and transient response of a plugged acoustic cavity to external excitation. The plug is an elastodynamic material subject to external excitation. The excitation of the plug is transmitted to the acoustic cavity. No internal sources of excitation are considered. A three-dimensional BEM model is developed for the analysis, both for the fluid in the cavity and the elastodynamic material in contact with it. Transient response is determined by recovering the time domain response from the Laplace transform domain solution using an FFT-based inversion scheme. Steady-state response is obtained from the Laplace transform domain solution by setting $s = -iw$. The numerical code is designed to run on the IBM family of personal computers and workstations.

EE2. Transmission loss studies of muffler and duct systems by the boundary element method. C. Y. R. Cheng and A. F. Seybert (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506-0046)

The present paper deals with transmission loss studies of acoustical muffler and duct systems using the boundary element method (BEM). Results obtained by using this approach are presented for both axisymmetric and fully three-dimensional models. The transmission loss of a muffler element can be computed using the transfer matrix approach. This method is ideally suited for the BEM because only the values of the unknowns at the inlet and outlet boundaries are required. The BEM values are compared to theoretical predictions and other available data such as finite element data, and good agreement is obtained within a broad frequency range. It can be concluded that the BEM should give reliable estimates of transmission loss in practical situations. [Work supported in part by ASHRAE.]

EE3. Finite difference modeling of acoustic pulse propagation in the atmosphere. Victor W. Sparrow (Department of Electrical and Computer Engineering, University of Illinois, Urbana-Champaign, Champaign, IL 61820 and USA-CERL, P.O. Box 4005, Champaign, IL 61820-1305) and Richard Raspet (Physical Acoustic Research Group, The University of Mississippi, University, MS 38677)

In a previous presentation [J. Acoust. Soc. Am. Suppl. 1 80, S104 (1986)], the use of a finite difference approximation of the time-dependent acoustic equations for arbitrary axisymmetric pulses was described. In that paper, however, only the application of these equations for a time harmonic simple source was demonstrated. In the presentation, the initial

condition requirements on the acoustic variables' values composing an acoustic pulse will be addressed and results for the numerically simulated propagation of such a pulse will be shown. Further, an extension to the above model for atmospheric propagation currently being tested will be described that includes both classical and molecular relaxation absorption effects. Space permitting, the implementation of our programs on a CRAY X-MP supercomputer will be commented upon.

EE4. An on going study of the use of the boundary element method to solve some of the more commonly encountered industrial noise and vibration control problems. Kassem M. Mourad (Department of Mechanical Engineering, North Carolina State University, Raleigh, NC 27695-7910), Robert D. Ciskowski (International Business Machines, Rochester, MN 55901), and Larry H. Royster (Department of Mechanical Engineering, North Carolina State University, Raleigh, NC 27695-7910)

A study is under way to establish the advantages or disadvantages of using the boundary element method (BEM) to obtain solutions to some of the more commonly encountered noise and vibration control problems encountered in general industry. Initial investigations include predicting the steady-state and transient room-volume SPL response from internal and boundary acoustic sources for various boundary impedances. The numerical code developed to obtain the solutions of such problems was designed to run on the IBM family of personal computers and workstations.

EE5. Plane-wave excitation of an infinite circular cylinder reinforced by a periodic set of rings. Francois De Maigret (Thomson-Sintra A.S.M., Route des Dolines, Parc de Valbonne, B.P. 38 06561, Valbonne Cedex, France)

Analytical expressions are derived for the scattered field by an infinite fluid-loaded cylindrical shell reinforced with the periodic set of rings and acoustically excited by an incident harmonic plane wave. Vibrations of the shell are treated using thin bending cylindrical shell theory by Kennard. The stiffeners interact with the cylindrical shell only through normal forces. Unknown quantities (i.e., scattered acoustic field, displacements of the shell) are expanded on the natural Bloch-Floquet basis induced by the structure periodicity. Numerical simulations on a wide range frequency are reported, where the dimensionless product "ka" of the incident fluid wavenumber by the shell radius typically extends from 10^{-2} to 10^2 , using a unique algorithm. Numerical predictions are compared with experimental data on different physical situations showing a good agreement. [Work supported by GERDSM.]

EE6. Effect of background noise on sound power estimates using the sound intensity technique. U. S. Shirahatti, Malcolm J. Crocker, and P. K. Raju (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

The presence of a background noise source affects the accuracy of the estimate of the source sound power level. In this paper, the results of a systematic experimental study on the accuracy of estimation of sound power levels in the presence of a background noise source are reported. A calibrated sound power source was used in these experiments. The sound power measurements were made using the two-microphone sound intensity technique. The errors in the estimates of the sound power level of the calibrated sound source were determined for different levels of background noise. Also, a systematic study was made on the relationship between the number of points of measurement on the enclosing surface defined around the source and the accuracy of the sound power estimate. It has been demonstrated that the accuracy of sound power estimates is greatly improved by continuous hand scanning of the enclosing surface instead of by making measurements at a small or moderate number of fixed points. The local sound pressure minus intensity index and the global sound pressure minus intensity index are used as indicators of the data quality. [Work supported by IBM, Charlotte, NC.]

EE7. Acoustic radiation from plates driven by multiple point random forces. H. Peng and R.F. Keltie (Center for Sound and Vibration, Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910)

The problem of acoustic radiation from plates driven by multiple random point forces is analyzed in this study. First, the formulation of the sound power radiated by infinite thin plates (one dimension) under the action of multiple random point forces is derived. The reaction of the acoustic medium on the vibratory response of plates is taken into account. With the input spectrum being assumed as bandlimited white noise, the effects of input force correlation on the radiated sound power level are examined. The analysis is then extended to finite thin plates with four simply supported edges. The fluid loading effects are neglected. The exact expressions for the surface acoustic intensity and the radiated sound power are also derived. Assuming light damping, approximate solutions for acoustic intensity and sound power are obtained. An experimental study is carried out to measure the surface acoustic intensity patterns and compare with the analytical results. The effects of input force correlation on the acoustic intensity patterns and the sound power level are discussed.

EE8. Prey detection by means of passive listening in bottlenose dolphins (*Tursiops truncatus*). Nélio B. Barros and Arthur A. Myrberg, Jr. (RSMAS-BLR, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149-1098)

Free-living bottlenose dolphins (*Tursiops truncatus*) are primarily piscivorous and the use of active echolocation has been implicated in prey capture [K. S. Norris and B. Møhl, *Am. Nat.* **122**, 85-113 (1983)]. Recent evidence based on the examination of stomach contents of more than 70 individuals from the southeastern United States has suggested that bottlenose dolphins also likely use passive listening for the purpose of detecting and possibly orienting to their prey. The most frequently occurring prey species (i.e., those found in more than 15% of the sample) were invariably those known to be highly active sound producers and, in all instances, producers of sounds of high level (e.g., croakers and drums of the family Sciaenidae, grunts of the family Haemulidae, toadfishes-midshipmen of the family Batrachoididae, and mullets from the family Mugilidae). Although in several instances members of some species could have been captured by active echolocation and visual means, it is highly unlikely that members of other species, by their highly secretive habits, would have been captured by such methods (i.e., members of the family Batrachoididae). Data from other regions corroborate the abovementioned findings that fish producing loud sound comprise the bulk of the diet of bottlenose dolphins.

EE9. Effect of a semicircular diffuser on the sound field in a rectangular room. Pan Jie and D. A. Bies (University of Adelaide, Department of Mechanical Engineering, G.P.O. Box No. 498, Adelaide, S.A. 5001, Australia)

Previous experimental work has indicated the influence of a rotating diffuser upon the diffusion of a reverberation field, upon the radiation impedance of a sound source in a reverberation room, and upon the boundary absorptions. Some interesting results of this work need a quantitative interpretation while others suggest an effective investigation to obtain a detailed physical insight. In an effort to find an insight into and an interpretation of this work, an analytical approach is presented. This paper is an interim report on continuing research, and it reviews an investigation of the effect of a semicircular diffuser upon the two-dimensional sound field in a rectangular room. The numerical relationship between the resonance frequencies of the acoustical modes in the room and the diffuser orientation is given. The sound-pressure distributions of the acoustical modes are also given as the function of the diffuser orientation. A three-dimensional cavity is used for experimental verification. In this cavity, only the sound waves in the horizontal plane will be affected by a diffuser. The experimental and numerical results agree closely.

EE10. The application of the cepstrum technique to the separation of the input signal and the structure response. Mei Q. Wu and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

In this paper, an iterative algorithm for the cepstrum is presented. The improved cepstrum technique is applied to separate the input signal from the response of a structure. It is proved that if the input signal is an impulse and the frequency response of the structure is bandlimited, then the frequency response function of the structure can be determined from the output of the structure without any preknowledge of the input signal. Furthermore, the spectrum of the input signal can be determined from the structure response function and the output spectrum. Some computer simulated experiments have been conducted. The experimental results show that the frequency response function recovered by using the iterative algorithm converges to the correct response function, and for about 30 times of iteration the recovered response function is very similar to the correct response function. [Work supported by IBM, Austin, TX.]

EE11. Subjective evaluation of a test earmuff exhibiting flat attenuation with nonlinear characteristics. June L. Peters (Department of Industrial Engineering, North Carolina State University, Raleigh, NC 27695), Larry H. Royster (Department of Mechanical Engineering, North Carolina State University, Raleigh, NC 27695), Julia Doswell Royster (Environmental Noise Consultants, Inc., Cary, NC 25712), and Richard G. Pearson (Department of Industrial Engineering, North Carolina State University, Raleigh, NC 27695)

Subjective user responses were obtained for a test earmuff exhibiting approximately flat attenuation of about 25 dB from 500-8000 Hz for SPLs less than 115 dB, with attenuation increasing nonlinearly up to about 10 dB at higher SPLs. The study population consisted of police officers performing annual gunfire requalification, in which officers execute required target shots during two sequential relay exercises. In the first phase of the study, subjects wore either their regular hearing protector or the test muff during the first relay, then the alternate protector for the second relay (with order counterbalanced). In the second phase, additional subjects wore either a control earmuff that was identical to the test earmuff except for the nonlinear mechanism (also eliminating the flat response characteristics) or the test earmuff. Subjects completed questionnaires concerning their opinions of the hearing protectors worn during both phases. Preliminary results indicate a significant difference in favor of the test earmuff in three areas: speech understanding, perceived level of protection, and comfort.

EE12. A generalized theory for acoustical holography and acoustical diffraction tomography. W. S. Gan (Acoustical Services Pte Ltd., 29 Telok Ayer Street, Singapore 0104, Republic of Singapore)

A generalized theory is proposed to combine acoustical diffraction tomography (ADT) with acoustical holography (AH) to take account of the diffraction and scattering during transmission of sound waves through an inhomogeneous medium. This is usually neglected in AH but must be considered if the medium is inhomogeneous. In this generalized theory, AH is a nearfield solution of the Helmholtz wave equation and ADT is a

farfield solution. The nonlinear case of strong scattering in the inhomogeneous medium is considered. The Feynman path integral is applied to the problem, and this can give the solution in closed form. Previous works [M. Slaney and A. C. Kak, paper at the 1985 IEEE Ultrason. Symp.; and Z. Q. Lu, IEEE Trans. Ultrason. Ferroelectrics and Frequency Control UFFC-33(6), 722-730 (1986)] cannot give the solution in closed form. The path integral is based on the action principle (variational principle). Here, the Schrödinger equation will be used. The scattering integrals, perturbation expansion of the scattering potential, and the wavefunction are all expressed in terms of the path integrals. The algorithm of the imaging process for the generalized theory is given.

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987

UNIVERSITY LECTURE HALL, 2:15 TO 3:15 P.M.

Session FF. Architectural Acoustics IV: Technical Committee on Architectural Acoustics

V. O. Knudsen Distinguished Lecture

William J. Cavanaugh, Chairman

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

Chairman's Introduction—2:15

Invited Paper

2:20

FF1. Designing for the performing arts: An historical overview. Michael Forsyth (Department of Drama, University of Bristol, 29 Park Row, Bristol BS1 5LT, United Kingdom)

A relationship developed historically between the acoustics of different auditorium types and corresponding styles of musical composition. This resulted from the composer's intuitive awareness of room acoustics, and from the architect's limited knowledge, based on trial and error, of the acoustic performance of building materials and room shapes in providing for different purposes. Except where composers have had auditoria specifically designed for their own music, this relationship became tenuous in the present century when the musical repertoire broadened and when nonmusical criteria came to dominate the design process. An auditorium's success now depends on the initial selection of acoustical and other criteria in relation to the building's defined purpose and on the translation of acoustic theory into three-dimensional form. These factors together determine the audience-performer relationship and the shape and dimensions of the enclosing surfaces. Historical examples illustrate early attempts to project theory into built form, and others from the present day describe the problem of integrating opposing acoustic demands.

Meeting of Accredited Standards Committee S2 on Mechanical Shock and Vibration

to be held jointly with the

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

J. C. Barton, Chairman S2

Caterpillar Tractor Company, Research Department, 100 N.E. Adams, Peoria, Illinois 61629

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

220 Clark Avenue, Brandford, Connecticut 06405

Standards Committee S2 on Mechanical Shock and Vibration. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interface of S2 activities with those of ISO/TC 108 (the Technical Advisory Group for ISO/TC 108 consists of members of S2, S3, and other persons not necessarily members of those committees) including plans for the next meeting of ISO/TC 108, to be held in September 1988.

WEDNESDAY AFTERNOON, 18 NOVEMBER 1987

UM AUDITORIUM, 3:30 TO 5:00 P.M.

Plenary Session

Chester M. McKinney, Chairman

President, Acoustical Society of America

Presentation of Awards

Distinguished Service Citation to Frederick E. White

Honorary Fellowship to Henrik A. S. Nødtvedt

Silver Medal in Psychological and Physiological Acoustics to Eberhard Zwicker

Silver Medal in Speech Communication to Dennis H. Klatt

Trent-Crede Medal to Miguel C. Junger

Local musical entertainment will conclude the session.

Session GG. Musical Acoustics I: Extended Vocal Techniques, and Production and Perception of Music

Gary L. Gibian, Chairman

Department of Physics and Audio Technology, American University, Washington, DC 20016

Chairman's Introduction—8:30

Invited Paper

8:35

GG1. Harmonic singing and the Harmonic Choir. David Hykes (The Harmonic Arts Society, 1047 Amsterdam Avenue, New York, NY 10027)

In harmonic singing, the singer emphasizes a selected upper harmonic of the vocal pulse. In this way, he can sing two notes at once, the fundamental and the selected harmonic. Members of the Harmonic Choir have developed techniques for the following effects: (1) The fundamental is constant while the selected harmonic varies; (2) the fundamental varies in a melody while the selected harmonic remains constant, leading to parallel harmony; (3) both the fundamental and the selected harmonic vary, either in converging or in diverging directions. The Choir together performs unaccompanied works which are composed but not scored. Different performances follow a common path but differ in many details.

Contributed Papers

9:20

GG2. Regional cerebral blood flow for singers and nonsingers while speaking, singing, and humming a rote passage. C. Formby (Departments of Communicative Disorders and Neurology, University of Florida, Gainesville, FL 32610) and R. G. Thomas (Department of Biometry, Emory University, Atlanta, GA 30322)

Two groups of singers ($n = 12, 13$) and a group of nonsingers ($n = 12$) each produced the national anthem by (1) speaking and (2) singing the words, and by (3) humming the melody. Regional cerebral blood flow (rCBF) was measured at rest and during each phonation task from seven areas in each hemisphere by the ^{133}Xe -inhalation method. Global, intrahemisphere, and interhemisphere rCBF were generally similar across phonation tasks and did not yield appreciable differences among the nonsingers and the singers. From these rCBF data, it was concluded that: (1) the normal production of a familiar passage by speaking, singing, or humming requires the interaction of both cerebral hemispheres more or less equally and (2) these tasks are relatively independent of musical training. [Research supported by NIH.]

9:35

GG3. Note coupling in Caribbean steel drums. D. Scott Hampton, Clifford Alexis, and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The acoustical behavior of Caribbean steel drums is characterized by strong coupling between the different note areas [T. D. Rossing, D. S. Hampton, and J. Boverman, *J. Acoust. Soc. Am. Suppl.* 1 **80**, S102 (1986)]. This makes it difficult to maintain tuning stability, especially in the harmonics of the various notes. An improved new design makes use of double grooves to separate the various note areas. By means of holographic interferometry, the note coupling was compared in two comparable double second (alto) drums, one with single grooves and one with double grooves. The drum with double grooves exhibits less internote coupling at high amplitudes, and better tuning stability.

9:50

GG4. Modal analysis of a Caribbean steel drum. Uwe J. Hansen (Department of Physics, Indiana State University, Terre Haute, IN 47809) and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Modes of vibration and coupling between adjacent note areas in a double-second steel drum are compared by several complementary techniques, including impact modal analysis, holographic interferometry, and recording sound spectra under varying conditions of damping. Each note has at least one overtone mode tuned to a harmonic of the fundamental frequency. Striking the A_4^+ note area, for example, excites not only a second harmonic A_2^+ in that same note area but also the strategically located A_3^+ and A_0^+ note areas adjacent to it. The coupling depends upon amplitude in a nonlinear way.

10:05

GG5. Physical correlates of perceptual similarity among synthesized approximations to selected targets. G. L. Gibian, D. R. Clements, E. N. Harnden, H. E. F. Williams, and R. K. Massaro (Physics Department, American University, Washington, DC 20016)

Experiments are described to evaluate the relative merits of several measures for predicting the perceived closeness of synthesized approximations to selected steady-state portions of musical tones ("targets"). Approximations were synthesized by means of audio-rate frequency modulation [Chowning, *J. Audio Eng. Soc.* **21**, 526-534 (1973)] using a computer program developed in a previous paper [Gibian *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S46 (1987), and *Audio Engineering Society Preprint #2380*]. The importance of including level- and frequency-dependent error weightings according to the Fletcher-Munson curves will be assessed. This information can be useful to composers who are blending electronic and traditional instruments in mixed ensembles.

10:20

GG6. Comprehensive study of analysis and synthesis of tones by spectral interpolation. Roger Dannenberg, Marie-Helene Serra, and Dean Rubin (Center for Art and Technology, Carnegie Mellon University, Pittsburgh, PA 15213)

A new approach to the real-time generation of digital sounds uses a completely automated analysis/synthesis technique for natural sounds. This approach leads to a more efficient implementation than classical additive synthesis; moreover it allows dynamic spectral variations to be controlled with only a few high-level parameters. Additive synthesis devices require a large number of oscillators (one for each partial). This technique gives excellent results; however, it requires a large amount of computation, and a large amount of control data. On the other hand, fixed-waveform synthesis uses only one oscillator, but the results are of poor musical quality since there is no dynamic evolution of the spectrum. A new technique has been investigated in which spectral variation is achieved through spectral interpolation. The research shows that spectral interpolation provides high-quality synthesis including controlled timbral variation at little more than the cost of a table-lookup oscillator. The task of analyzing different kinds of instrumental sounds to produce control information for this technique has been automated.

10:35

GG7. Nonuniformity in timbre of string instruments. Asbjørn Krokstad (Division of Telecommunications, The Norwegian Institute of Technology, N-7034 Trondheim, Norway)

Mechanical musical instruments, may show great variation of all physical properties that correlate to timbre, even over intervals of a semitone. To examine preferences and limits of acceptability, timbre variations have been studied in a melodic context. A short melody of 13 tones covering a decime was played by two professional musicians on three violins and a viola, and recorded at two microphone positions in an anechoic room. Each tone was processed digitally to reduce differences in loudness and in transients. In the first of two listening tests, dissimilarities in timbre between pairs of successive tones were evaluated. From these differences, integrated differences for different versions of the whole melody are calculated, and differences between versions. The possibility of identifying a certain player, a certain instrument or a certain microphone position may be given as relative numbers. The dissimilarity data are used for the synthesis of new versions of the melody with a range of nonuniformity in timbre. In the second listening test, acceptability of nonuniformity were evaluated. To study the dimensions and physical variabilities of timbre by sustained tones, multidimensional analysis of the subjective differences are performed and several spectral variables for describing the differences are tested.

10:50

GG8. Theoretical relationship between bow pressure and vibration amplitude, derived from a sticking-sliding force-velocity characteristic. Xavier Boutillon and Gabriel Weinreich (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

The Helmholtz motion for bowed strings may be described with the help of a characteristic between the bow's frictional force and the string velocity at the bowing point. Although this characteristic has never been dependably measured, some of its features are implied by players' common experience. In general, the vibration amplitude changes with bow pressure or bow velocity. The relationships between these changes and the local evolution of the characteristic will be established and discussed. The case of perfect stickiness, in which the amplitude becomes independent of

bow pressure, follows as a special case. [Work supported in part by NSF and the French Ministry of Culture.]

11:05

GG9. Impedance characteristics above cutoff for clarinet-like structures. William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

The cutoff frequency determined by the open tone holes on a clarinet-like structure is a useful means for characterizing the structure's musical properties. [See, for example, the discussion and data in A. H. Benade, *Fundamentals of Musical Acoustics* (Oxford U.P., London, 1976), Chaps. 21-22; note Sec. 22.5 in particular.] When impedance peaks above the cutoff frequency have amplitudes that are significant fractions of those below cutoff, they may influence the clarinet reed vibration. The input impedance of a clarinet-like structure (a cylindrical tube in the present case) is influenced by its input and output terminations. Three input terminations are considered: (1) rigid cap, (2) rigid taper, and (3) rigid taper with reed. Several output terminations are considered including (1) open end, (2) "infinite" tone hole lattice, (3) "short" tone hole lattice, and (4) short lattice plus bell. Input impedances are calculated for various combinations of the input and output terminations to demonstrate their effects on the amplitudes of the input impedance peaks both above and below the cutoff frequency.

11:20

GG10. Spectral characteristics of the Helmholtz resonator driven by an air jet. Roset Khosropour and Peter Millet (Department of Physics, Hamilton College, Clinton, NY 13323)

Motivated by an interest in the action of air jets on resonant systems in woodwinds, preliminary measurements have been made on the frequency spectra of several air jet driven Helmholtz resonators. Over the range of air jet velocities examined, stable oscillations dominated by a single frequency (under certain conditions harmonics are in evidence) occur in several intervals of jet velocity. Within each such interval the dominant frequency increases with increasing jet velocity. The span of frequencies associated with each interval includes the calculated Helmholtz frequency and, depending on the resonator and the interval, can be as much as an octave. The spectrum of the zone between the intervals is formed from an overlap of the extrapolated spectra of the adjacent intervals, i.e., it is marked by the appearance of two frequencies, where one is above the maximum frequency from the lower interval and the other is below the minimum frequency from the higher interval.

11:35

GG11. Acoustical behavior of a bass drum. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The bass drum is one of the loudest instruments in the orchestra. Its two large membranes may be set to equal tension, but more often the batter head is tuned to a higher tension. The sound spectrum contains many peaks that can be related to the vibration modes of a membrane. The low tension of the heads plus the strong coupling via the enclosed air lead to a large frequency difference (about 1½ octave) between the two (01) modes associated with in-phase and out-of-phase motion of the heads. Sound decay rates for most modes are slightly greater when the two heads are at the same tension.

Session HH. Psychological and Physiological Acoustics VI: External Ear, Inner Ear, and Cochlear Implants

William W. Clark, Chairman

Central Institute for the Deaf, 818 S. Euclid, Saint Louis, Missouri 63110

Contributed Papers

8:30

HH1. Static and dynamic compliances of the rat tectorial membrane under *in vivo*-like conditions. Glenn H. Frommer (DAMPA A/S, 5690 Tommerup, Denmark)

Compliances were examined qualitatively in 17 rat cochleas prepared minutes after death. Forces were applied perpendicularly to the membrane by a 10- μ m-thick microelectrode that could also vibrate. Sinusoidal displacements were 10- μ m peak to peak. White and strobe light illumination was used [G. H. Frommer, *Acta Oto-Laryng.* **94**, 451-460 (1982)]. Static depression of the membrane at points between the spiral sulcus to the outer hair cells produced an elongated hollow, while almost circular depressions were noted in the lip region. Dynamic characteristics were frequency dependent. Under 60 Hz, depression patterns were similar to the static case. However, at 200 Hz, the region between the sulcus and outer hair cells was quite rigid and displaced at the limbal attachment. Local vibrations of the membrane in the lip region and over the sulcus were almost circular. The mechanical characteristics correlate with ultrastructural studies. It is concluded that static analyses alone cannot describe the viscoelastic behavior of the tectorial membrane.

8:45

HH2. Stiffness of the tectorial membrane in live Mongolian gerbils. J. J. Zwislocki, M. W. Serafini, and L. K. Cefaratti (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290)

The classical model of hair-cell stimulation in the mammalian cochlea, according to which the tectorial membrane acts as a stiff anchor for the cells' stereocilia, requires the stiffness of the membrane in the radial cochlear direction to be greater than that of the stereocilia. This stiffness was measured by means of flexible, calibrated micropipettes inserted deep into the membrane at the location of the outer hair cells. They were introduced into the cochlea through an opening in the lateral wall of the scala media, as described previously [J. J. Zwislocki, S. C. Chamberlain, and N. B. Slepceki, *J. Acoust. Soc. Am. Suppl.* **181**, S6 (1987)]. The stiffness was determined by moving the micropipettes radially and by measuring their bend and the resulting displacement of the tectorial membrane. According to the experiments, the radial stiffness of the tectorial membrane is several times smaller than the corresponding stiffness of the stereocilia determined by Strelhoff and Flock (*Hearing Res.* **15**, 19-28). This contradicts the classical model of cochlear hair-cell stimulation. The tectorial membrane appears to act on the stereocilia as a mass load rather than their stiff anchor.

9:00

HH3. Three-dimensional analysis of fluid flow between the tectorial membrane and the organ of Corti. Joseph A. M. Boulet (Department of Engineering Science and Mechanics, University of Tennessee, Knoxville, TN 37996-2030) and J. Barney Holt (Space Transportation Systems Division, Rockwell International, 555 Discovery Drive, Huntsville, AL 35806)

Three-dimensional solutions for propagating waves in the fluid-filled gap between the tectorial membrane (TM) and the organ of Corti (OC) are obtained and included in an existing analytical model of the cochlea. The three-dimensional solutions give essentially the same coupling between TM and OC as that previously found with two-dimensional solutions. However, including the three-dimensional solutions gives rise to a new propagating mode. With two-dimensional waves in the gap, the model exhibits three ducts (two scalae and the inner sulcus) and two propagating modes. With three-dimensional solutions, the gap functions as a fourth duct. Consequently, a third propagating mode appears. The flow patterns associated with this mode are described, and the mode's significance is discussed [Work supported by NSF.]

9:15

HH4. Quantification of cerebrospinal fluid (CSF) contamination of scala tympani (ST) perilymph samples. Akira Hara, Alec N. Salt, Jogy Varghese, and Ruediger Thalmann (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

Previously it has been reported that levels of most amino acids in small perilymph samples (< 100 nL) taken from ST through the round window (RW) of guinea pigs are distinctly different from those in CSF [A. N. Salt and R. Thalmann, *Adv. Otorhinolaryngol.* (in press)]. The composition of samples of larger volume more closely resembles CSF. Now an analysis of amino acid content has been performed as the sample volume withdrawn through the RW was systematically varied from 100 nL to 5 μ L. Glycine, serine, glutamine, and alanine were measured by HPLC (Physiological Pico-Tag, Waters). Up to 200 nL, the glycine content of ST perilymph was similar to that of scala vestibuli. With larger volumes, the glycine content rapidly declined toward that of CSF. These data suggest that perilymph samples taken through the RW are contaminated with CSF if more than 200 nL is withdrawn. If ST perilymph is sampled through a small fenestra in the third turn, larger volumes (up to 2 μ L) can be withdrawn without significant CSF contamination, evidently because of the distance from the cochlear aqueduct. The data also suggest that glycine content can be used as an indicator of CSF contamination of ST perilymph samples.

9:30

HH5. Two-microphone measurement of the power-reflection coefficient in human ears. William J. Murphy, Arnold Tubis (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)

For each of eight normal hearing and eight abnormal hearing subjects, the acoustic intensity and power reflection coefficient spectra are measured using a two-microphone probe with probe tips \approx 8 mm downstream from the point of sound delivery [W. J. Murphy, A. Tubis, and G. R. Long, *J. Acoust. Soc. Am. Suppl.* **181**, S75 (1987)]. The tube of an Etymotic Research ER-2 sound delivery system passes through one port and the probe tips from a pair of ER-7 microphones pass through the second port of a latex ear mold. Results in the frequency range of 4-12

kHz are compared with previous measurements by Stinson *et al.* [*J. Acoust. Soc. Am.* **72**, 766–773 (1982)], Hudde [*J. Acoust. Soc. Am.* **73**, 242–247 (1983)], and Stinson [*J. Acoust. Soc. Am.* **77**, 386–393 (1985); **79**, 1003–1009 (1986)]. Hearing thresholds are also determined in order to identify any significant correlations in the objective and psychophysical measure. [Work supported by a grant from the Deafness Research Foundation.]

9:45

HH6. The relation between the impedance at the microphone and the transfer impedance of an occluded ear simulator. George F. Kuhn (Vibrasound Research Corporation, 2855 West Oxford Avenue, Englewood, CO 80110)

The ANSI S3.25-1979 Standard for an occluded ear simulator specifies the transfer impedance, that is the ratio of the pressure at the measuring microphone to the volume velocity 12.7 mm in front of the microphone, from 100–10 000 Hz. This paper presents analytical relationships between the transfer impedance and the simulated eardrum impedance. Limit of error calculations are used to derive the expected tolerances for one impedance when the tolerances are specified for the other impedance. [Work supported by NIH, NINCDS.]

10:00

HH7. Longitudinal resonances of the human, external ear: A first approximation. George F. Kuhn (Vibrasound Research Corporation, 2855 West Oxford Avenue, Englewood, CO 80110)

The characteristic equation for a simple model of the external ear is used to demonstrate the individual effects of the radiation impedance, of the pinna, and of the ear canal, including a wedge-shaped termination at the tympanic membrane, on the resonance frequencies of the ear. The results show that a denotation such as “first quarter wave resonance,” suggesting a physical dimension of the ear, which is a quarter wavelength long at the resonance frequency, is a misnomer. [Work supported by NIH, NINCDS.]

10:15

HH8. Absorption, phase velocities, characteristic impedances and transmission matrices for tubes of small diameter. George F. Kuhn (Vibrasound Research Corporation, 2855 West Oxford Avenue, Englewood, CO 80110)

Numerical results for the absorption, the phase velocities, the characteristic impedances and the transmission matrices, relating input pressure and velocity to output pressure and velocity, are presented for tubes with small radii relative to the boundary layer thickness. These exact results are compared to results based on other approximate analytical models and their range of validity in terms of frequency and tube radius is examined. The exact analytical model yields electroacoustic analogs, for certain sizes and frequencies, which contain nonrealizable negative resistances. It is shown that a change in the length of the tube(s) comprising the transmission line overcomes this difficulty. [Work supported by HH, NINCDS.]

10:30–10:45

Break

10:45

HH9. A receptor-coded cochlear implant: Speech discrimination and comprehension results. Gerald S. Wasserman (Department of Psychological Sciences, Purdue University, W. Lafayette, IN 47907) and Richard T. Miyamoto (Department of Otolaryngology, Indiana University Medical School, Indianapolis, IN 46202)

Cochlear implants are, among other things, artificial receptors. Like hair cells, they transduce acoustic energy into bioelectric signals that activate auditory nerve fibers. The contribution of receptor coding to perception in adult and juvenile deaf patients surgically fitted with cochlear implants has therefore been investigated. A filter-feedback system was used to simulate the signal processing of natural receptors. This system permitted controlled within-patient tests of the effect of receptor coding on speech perception; this effect was examined in two different ways: Speech discrimination was tested with CVC monosyllables that differed only in their final phoneme. Speech comprehension was tested with a modification of the SPIN test. All tests were run as two-alternative forced-choice tasks. Patients were completely alone in a windowless acoustic booth during testing; stimulation and data collection were under computer control. In every case, receptor coding produced a significant improvement in phoneme discrimination. The sentence comprehension task was more difficult and most patients were at or near chance in both experimental and control conditions; however, receptor coding can produce clearly significant performance in a patient whose control sentence comprehension is at chance.

11:00

HH10. Vowel discrimination using different frequency-to-electrode maps in cochlear implant users. Mario A. Svirsky (Kresge Hearing Laboratory of the South, ENT Department, LSUMC, New Orleans, LA 70112 and Department of Biomedical Engineering, Tulane University, New Orleans, LA 70118), John K. Cullen, Howard P. Ragland (Kresge

Hearing Laboratory of the South, ENT Department, LSUMC, New Orleans, LA 70112), and Cedric F. Walker (Department of Biomedical Engineering, Tulane University, New Orleans, LA 70118)

The Nucleus WSP-III is a speech processor for cochlear implants that stimulates two electrodes per fundamental period. The electrodes stimulated are chosen based on the first two formant frequencies: lower frequency formants excite electrodes that are more apical. This gives users a cue to discriminate vowels with different formants. But, if two particular vowels had second formants close enough to fall within the frequency band of some electrode, the user would miss that cue—those two vowels would “behave” as if they had exactly the same second formant. Cochlear implant users were tested under three frequency-to-electrode maps: the standard map (used by most Nucleus implant patients), the modified standard map (proposed by P. Blamey, private communication, 1987) and a new map designed to maximize contrast between different vowels. A discrimination test was used where subjects were not asked to identify the sounds they heard, but only to indicate whether the different word in a set of three was in the initial or final position. This tends to minimize learning effects due to prior experience with a given map. Also, if cochlear implant users can discriminate different phonemes, they could conceivably be trained to identify them correctly, whereas lack of discrimination makes identification impossible. [Work supported by Cochlear Corp. and Sigma Xi.]

11:15

HH11. Evaluation of a multichannel cochlear implant. Sigfrid D. Soli, Virginia M. Kirby, Christopher van den Honert, and Gregory P. Widin (Hearing Research Laboratory, 270-4S-11, 3M Center, Saint Paul, MN 55144)

A wearable multichannel signal processor for stimulation of single-electrode cochlear implants has been field tested with two patients. Each channel in the processor, which is implemented in a digital signal processing chip, consists of a resonator followed by an instantaneous compressive nonlinearity. The channel outputs are digitally mixed for use with single-electrode implants. The resonators perform a spectral-to-temporal transformation of the input signal and the nonlinearities limit output level to emulate the response characteristics of normal auditory neurons. The resonator and nonlinearity parameters are adjusted to accommodate both the acoustic properties of speech sounds and the electrical dynamic range of the patient. Several processor configurations with different resonator and nonlinearity designs have been evaluated. The results of psychophysical tests, used to fit each processor configuration to the patient and measure speech performance in quiet and noise with each configuration, will be reported.

11:30

HH12. Backward and forward masking for direct electrical stimulation of the VIIIth nerve in two profoundly deaf subjects. L. J. Dent and B. S. Townshend (Stanford Electronics Laboratories, Stanford, CA 94305)

Two profoundly deaf multielectrode implant subjects were required to detect a probe signal (10 ms in duration) in a temporal gap between two pulse-train maskers (each 300 ms in duration). The detection threshold was measured for a probe centered temporally in the gap, as well as for a probe offset from center by up to 97.5%. Also presented were the pure backward and pure forward masking cases. Qualitatively, both subject's forward and backward masking functions approximated those observed for normal hearing subjects [L.L. Elliott, *J. Acoust. Soc. Am.* **34**, 1116-1117 (1962)] in that forward masking decayed more gradually than backward masking as a function of probe-masker separation. Because mechanical (cochlear) contributions to masking [H. Duifhuis, *J. Acoust. Soc. Am.* **54**, 1471-1488 (1973)] can be excluded in the case of direct VIIIth nerve stimulation, these data support the attribution of nonsimul-

taneous masking phenomena to VIIIth nerve or higher neural mechanisms. [Work supported by NIH.]

11:45

HH13. Channel interactions measured by forward-masked "place" tuning curves with multichannel electrical stimulation. Virginia M. Kirby (Hearing Research Laboratory, 270-4S-11, 3M Center, Saint Paul, MN 55144), David A. Nelson (Hearing Research Laboratory, University of Minnesota, Minneapolis, MN 55455), Sigfrid D. Soli (Hearing Research Laboratory, 270-4S-11, 3M Center, Saint Paul, MN 55144), and Todd W. Fortune (Hearing Research Laboratory, University of Minnesota, Minneapolis, MN 55455)

Simultaneous stimulation of multichannel intracochlear electrodes can give rise to peripheral and central channel interactions. The ability to eliminate or predict and control the interactions produced by a given electrode geometry is a processing goal for optimizing performance with a multichannel cochlear implant. Previous studies have used loudness summation and forward-masking pattern techniques to estimate interactions between channels of electrical stimulation. In this study, interactions between bipolar channels of analog electrical stimulation were estimated using a forward-masking paradigm with a fixed-level, fixed-location probe. By varying the electrode location of a 200-Hz, 300-ms sinusoidal masker and determining the level of the masker at each location necessary to just mask a 200-Hz, 10-ms probe, a "place" tuning curve was derived. The level of masker required at a given location to mask the probe depends on the amount of excitation produced by the probe and reflects, in part, the degree to which there is overlap of neural populations responding to each stimulus. These "place" tuning curves, which display interactions as a function of masker location were determined for several probe levels and probe locations. Results and implications for speech processing strategies will be discussed.

THURSDAY MORNING, 19 NOVEMBER 1987

UMS 4 AND 5, 8:30 A.M. TO 12:00 NOON

Session II. Underwater Acoustics VI. Signal Processing for Underwater Acoustics (Précis-Poster Session)

Thomas G. Muir, Chairman
NATO SACLANT Centre, La Spezia, Italy

Chairman's Introduction—8:30

Contributed Papers

Following presentation of the précis, posters will be on display until 12:00 Noon.

8:35

III. Nonorthogonality of measured normal modes in shallow water. Grayson H. Rayborn (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004 and Department of Physics and Astronomy, University of Southern Mississippi, Hattiesburg, MS 39406), George E. Ioup, and Juliette W. Ioup (NORDA, NSTL, MS 39529 and Department of Physics and Geophysical Research Laboratory, University of New Orleans, New Orleans, LA 70148)

The importance of surmounting the nonorthogonality of measured normal modes and processing shallow-water data in such a way that mod-

al compositions are effectively recovered for matched field processing has been demonstrated by several investigators. The potential for improvement using this technique is greatest when the nonorthogonality of the measured modes is largest. The amount by which the normal modes fail to be orthogonal for a variety of ocean bottoms, array lengths and discretizations, and array positions for the Pekeris model has been studied. Environments are selected to reflect sediment types characteristic of the continental shelf. It has been found that the nonorthogonality is greatest for water depths and frequencies at and just above modal onsets, for sound-speed ratios close to 1, and for arrays which span only a fraction of the water column. Judicious placement and length selection for short arrays, however, can give orthogonal measured modes for some combinations of frequencies and environments when a small number of modes are present.

II2. Effects of measured mode nonorthogonality on conventional matched field processing. Juliette W. Ioup, George E. Ioup (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004 and Department of Physics, University of New Orleans, New Orleans, LA 70148), Grayson H. Rayborn (NORDA, NSTL, MS 39529 and Department of Physics and Astronomy, University of Southern Mississippi, Hattiesburg, MS 39406), Donald R. Del Balzo (NORDA, NSTL, MS 39529), and Christopher Feuillade (ODSI Defense Systems, Inc., 6110 Executive Boulevard, Suite 320, Rockville, MD 20852)

As discussed by Rayborn *et al.* (preceding abstract), the potential improvement achievable from the use of modal filtering in matched field processing is greatest when the measured modes are least orthogonal. To assess the ability of matched field processing using a conventional cross-correlation estimator to realize this potential, ambiguity surfaces for a Pekeris waveguide have been constructed employing (a) the pressure fields at the hydrophones and (b) the amplitudes resulting from modal filtering. The quality of the surfaces is quantified utilizing measures that compare the height of the source peak above the mean and the height of the source peak above the standard deviation of the surface. The results indicate that modal filter processing, for a given array of hydrophones, offers the most improvement when bottom depth, bottom type, and frequency combine to produce measured modes that are the least orthogonal, and yields no improvement when the measured modes are orthogonal.

8:43

II3. Effects of correlated noise on the cross-spectral matrix in modal composition space. George B. Smith, Christopher Feuillade (ODSI Defense Systems, Inc., 6110 Executive Boulevard, Rockville, MD 20852), and Donald R. Del Balzo (Naval Ocean Research and Development Activity, Code 244, NSTL, MS 39529-5004)

Computer simulations of hydrophone cross-spectral matrices in a shallow-water waveguide were generated for signals with different mixtures of correlated and white noise. These matrices were then mapped to modal composition space, a space populated by vectors whose elements are the amplitudes for the trapped modes. It was found that, in modal composition space, the cross-spectral matrix is not sensitive to the difference between correlated and white noise, but is sensitive to the difference between noise and signal. While the distribution of signal and white noise among the elements of the cross-spectral matrix is similar before and after the mapping to modal composition space, the distribution of correlated and white noise is not. Temporally discrete noise sources, which are correlated at the hydrophones, but not from sample to sample, make little or no contribution to the off-diagonal elements of the cross-spectral matrix in modal composition space. This fact has significant implications for matched field processing in low signal-to-noise situations.

8:47

II4. Adaptive beamforming or matched field processing in media with uncertain propagation conditions. A. B. Baggeroer, H. Schmidt (Massachusetts Institute of Technology, Cambridge, MA 02139), W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375), and E. K. Scheer (Woods Hole Oceanographic Institute, Woods Hole, MA 02543)

Adaptive beamforming or matched field processing provides high resolution with sidelobe control if accurate replica fields can be generated. The generation of these replica fields is a formidable problem requiring knowledge of the complex ocean propagation environment. On the other hand, the detection problem in the ocean may not require high resolution, whereas sidelobe control is still an important issue. Relaxing resolution requirements suggest that a certain tolerance incorporating uncertainty of the propagation conditions is permissible, or even desirable, because of the difficulties both in specifying the medium exactly and in identifying global

peaks. This possibility of lowering the requirements on our knowledge of the environment is investigated with two methods: (1) by constructing multiple beam (constraint) algorithms and (2) by considering stochastic blurring by the medium. These two approaches are applied to plane-wave beamforming and matched field processing.

8:51

II5. Spatial matched processing for multipath propagation. Matthew Dzieciuch and T. G. Birdsall (Communication and Signal Processing Laboratory, 4242 EECS Building, Department of Electrical Engineering and Computer Science, The University of Michigan, Ann Arbor, MI 48104)

Underwater acoustic propagation is characterized by multipath or multimode propagation. Ray theory and mode theory are not fully adequate for modeling physical reality. Impulse responses can be more accurately calculated using Gaussian beam theory. Signal processors can be designed to take advantage of the channel complexity if the propagation is actually known so that detectability is increased. The proposed technique, channel matched filtering, synthetically backpropagates the wave front to a hypothesized source location. Accurate passive estimates of source location can be made without knowledge of the signal characteristics. GB theory can easily accommodate a range-dependent deep water environment. [This research supported by the Office of Naval Research.]

8:55

II6. A new technique of acoustic mode filtering in shallow sea. Harish M. Chouhan and G. V. Anand (Department of Electrical Communication Engineering, Indian Institute of Science, Bangalore 560 012, India)

A new technique of filtering acoustic normal modes, which overcomes many of the drawbacks of the earlier techniques, is presented in this paper. It is based on the fact that each normal mode in an isovelocity channel comprises a pair of plane waves with characteristic directions of propagation symmetrically disposed with respect to the channel axis. A vertical array of equispaced hydrophones is shaded so as to steer nulls in the directional response of the array along the directions of arrival of plane waves corresponding to the unwanted modes. All the shading coefficients are real, leading to simplicity in the hardware realization of the processor. The shading coefficients are invariant to a shift in the position of the array. The array need not span the entire depth of the ocean. Efficient filtering is possible even when the eigenfunctions of the modes have a significant penetration into the bottom. [Work supported by DOE, Government of India.]

8:59

II7. Matched catastrophe deconvolution with application to the inversion of marine seismic refraction data. Michael G. Brown (Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149) and Paul E. Bullwinkel (Applied Measurement Systems, Inc., 1415 S.W. 21st Avenue, Fort Lauderdale, FL 33312)

The problem of extracting accurate estimates of the travel times of unresolved arrivals (e.g., the second and third arrivals within a triplication) from a set of noisy bandlimited measurements of a time-dependent acoustic wavefield is addressed. The method of solution presented is based on the assumption that the underlying caustic structure (or, equivalently, the travel time curve structure) of the wavefield is known. Because generic caustics associated with causal wavefields take on only certain forms, this is a weak assumption. Additionally, it is assumed that the medium

structure varies as a function of depth only. This further restricts the class of observable caustic sections. These constraints are incorporated into a matched field processing approach to the deconvolution problem. The technique is demonstrated using a measured marine seismic refraction data set. A Herglotz-Wiechert inversion of the resulting kinematic data $T(R)$ agrees well with previously published waveform inversions of the same data set. Advantages of the new approach over linearized waveform inversions are discussed.

9:03

II8. Effects of nearby soft screens on idealized array reception. Rudolph Martinez and Ann Westgard (Cambridge Acoustical Associates, Inc., 54 CambridgePark Drive, Cambridge, MA 02140)

Arrays of sensors must often function in the neighborhood of diffracting and reflecting surfaces; these, in turn, represent effective passive sources that add to the incoming signal. The result is a computed wave-number spectrum contaminated by the presence of spurious targets in which the true sea source may appear shifted off its actual azimuth. Here, such effects are demonstrated for the fully analyzable situation of an unshaded array near a thin semi-infinite soft screen, taken either as a colinear aft baffle or a perpendicular backstop to the incident wave. Exact and asymptotic expressions have been derived to predict perceived strengths of virtual targets and associated azimuthal distortions; their reported numerical evaluations could serve as benchmark results against which one might gauge the effectiveness of sophisticated processing methods under similar array installation conditions.

9:07

II9. Limitations to source description set by a finite receiving aperture. Philip L. Stocklin (Consulting physicist, 439 Blue Jay Lane, Satellite Beach, FL 32937), and Norma L. Stocklin (Systems Analyst, 439 Blue Jay Lane, Satellite Beach, FL 32937)

Spatial sampling function theory results are stated for a monochromatic acoustic field and for a bandwidth-limited acoustic field. These results are applied to a finite receiving aperture to estimate the physical limitations to source descriptions obtainable from the outputs of elements of the aperture. Effects of variations in propagation conditions, received signal bandwidth, received SNR, and available processing time are shown. Relationships to inverse field processing are developed.

9:11

II10. Improved time delay estimates of underwater acoustic signals using beamforming and prefiltering techniques. Brian G. Ferguson (Weapons Systems Research Laboratory, Defence Science and Technology Organization; Royal Australian Navy Research Laboratory, P.O. Box 706, Darlinghurst 2010, Australia)

The cross-correlation method for estimating the differential time delay for an underwater acoustic signal to arrive at two spatially separated receivers is described. The output amplitudes of the two receivers are cross correlated and the lag time at which the cross-correlation function peaks provides an estimate of the time delay. Often the time delay estimates of the signal are corrupted by the presence of noise. By replacing each of the (omnidirectional) receivers with an array of receivers and then cross correlating the beamformed outputs of the arrays, it is shown that the effect of noise on the time delay estimation process is substantially reduced. Both conventional and adaptive beamforming methods are implemented and the advantages of array beamforming (prior to cross correlation) are highlighted using both modeled data (for tutorial purposes) and experimental data. The performance of the cross correlator is further

improved by using various prefiltering techniques to process the experimental data. Each of the prefilters—Hannan-Thomson, smoothed coherence transform, and phase transform—consists of a frequency-domain weighting function that is applied to the cross-spectral density function before the cross-correlation function is computed using the inverse Fourier transform. The prefilters are observed to enhance the estimation of the time delay by reducing the ambiguity associated with detecting the peak value of the basic (unweighted) cross-correlation function, which is oscillatory in nature.

9:15

II11. Stepped FM signals for acoustic remote sensing of geoacoustic properties of bottom sediments. Harry A. DeFerrari and Hien B. Nguyen (Department of Applied Marine Physics, R.S.M.A.S., University of Miami, Miami, FL 33149)

Signals used for sub-bottom profiling are usually high-intensity transients generated by explosions, air guns, sparkers, etc. Each transient has a unique complex frequency spectrum and serial transmissions cannot be coherently averaged. Coherent acoustic signals generated by electromechanical transducers are not widely used, owing to the relatively low intensity or bandwidth limitations, but even weak signals can be averaged for long periods if the propagation medium is stable, to produce very high equivalent signal levels. Likewise, it appears possible to synthesize very broadband signals by coherently averaging narrow-band signals. Several narrow-band signals transmitted serially can be processed as though they were transmitted simultaneously to synthesize signals of any desired bandwidth—a method called stepped FM. A system has been developed and tested for transmitting and receiving pseudorandom code stepped FM signals. Signals with a bandwidth of 1700 Hz and intensity levels greater than 210 dB after processing are used to probe the first few tens of meters of the bottom sediments. The resulting pulse duration, approximately 0.5 ms, allows for a separation of reflected from refracted paths. Travel times of refracted rays can then be used for tomographic inversions to yield compressional velocity profiles of the bottom sediments.

9:19

II12. Signal design for Doppler processing. Ziad Haddad and Bowen Parkins (AT&T Bell Laboratories, 14A420, Whippany Road, Whippany, NJ 07981)

The Doppler sonar processing requires that the illuminating signal have a spectrum with low sidelobes over the range of frequencies at which Doppler shifts are expected. This is conventionally achieved by "windowing" the driving sinewave. Uncertainties in the amplitude response (threshold onset and linearity) of typical transducers make it desirable to have alternate means of obtaining low sidelobes. One approach is to transmit a series of nonuniformly spaced short pulses of constant duration and amplitude. Pulse start times are specified to yield the desired spectral shape using the approach of A. Ishimaru [IEEE Trans. Ant. Propag. AP-10, 691-702 (1962)] developed for space tapering antenna array elements. The resulting spectrum is not affected by nonzero threshold onset or limited linear range. The sidelobe reduction that can be achieved depends on the number of pulses, hence, on the amplifier bandwidth. Amplitude shading and varying pulse durations allow the design of a second type of signal, a windowed cw signal that does not depend on threshold onset and is less sensitive to amplifier bandwidth. The sidelobe reduction achievable with this second design depends on the linear range.

9:23

II13. Phase recovery and calibration with underwater acoustic arrays. Nathan Cohen (Metropolitan College, Boston University, Boston, MA 02115)

Underwater acoustic imaging provides unique challenges because of the nature of the medium and difficulties in accurate measurement of

array observables. Baseline uncertainties and loss of phase locking make phase observables especially difficult to acquire in some cases. Here, linear and nonlinear imaging techniques that aid in the calibration and/or recovery of phase observables and the self-calibration of amplitude observables are described. Enhanced dynamic range and image accuracy are possible with these post-processing techniques, which include closure relations and phase referencing. Examples of the application of the techniques in other imaging disciplines, such as optics and radio astronomy, are discussed.

9:27

II14. Interpretation of coherence estimates determined for time series of limited length. Allen E. Leybourne, III (Department of Engineering Technology, University of Southern Mississippi, Box 5172 S. Station, Hattiesburg, MS 39406-5172)

Coherence of multichannel time series data has been calculated by the method of G. C. Carter and J. F. Ferrie [*A Coherence and Cross Spectrum Estimation Program, Programs for Digital Signal Processing* (IEEE, New York, 1979), No. 2.3-1-18]. The properties of this statistic has been contrasted through simulation studies at known noise levels for short and long time series through a range of 40 data segments. Results at less than eight data segments are shown to be difficult to interpret; however, modest improvements in the coherence estimate can be made by adjustments indicated by these studies. A method of calculating coherence estimates through a technique referred to as frequency bin averaging is described. The results obtained are contrasted with the conventional method of Carter. Although reliable estimation of coherence for short time series data (transient data) is difficult to obtain in the conventional sense, the comparisons made should provide a basis for interpretation of coherence estimates as well as suggest corrections that may be applied to relatively short time series sequences. [Work supported by NORDA.]

9:31

II15. The effect of multipath and sensor location offsets on the performance of a cross-correlation system. P. Bilazarian (Raytheon Company, Submarine Signal Division, 1847 West Main Road, Portsmouth, RI 02871)

The combined influence of multiple ray arrivals and sensor location offsets on the performance of a broadband cross-correlation system is investigated. The system consists of three sensors that are nominally positioned on a horizontal line. Source location parameters are determined from time-delay estimates obtained by cross-correlation techniques. An analytical model for the prediction of localization bias errors due to multipath and to offsets from nominal sensor positions is discussed. This model is applicable to an oceanic medium with a depth-dependent sound-speed profile. Results are presented for a variety of multipath conditions, such as those associated with direct path, surface duct, bottom bounce, and convergence zone types of propagation. The sensitivity of localization errors to variations in environmental parameters, such as sea state or bottom-reflection properties, is also investigated. It is demonstrated that multipath, especially when combined with sensor position offsets, can have a significant impact on system performance.

9:35

II16. A fast bistatic reverberation and systems model. D. J. Kewley (Naval Underwater Systems Center, New London, CT 06320 and Weapons Systems Research Laboratory, DSTO, Department of Defense, GPO Box 2151, Adelaide, SA, Australia), and H. P. Bucker (Naval Ocean Systems Center, San Diego, CA 92152)

The performance prediction of bistatic active sonar systems requires

the calculation of the reverberation and noise background along with the signal propagation characteristics. To consider the general case of a three-dimensional beam pattern for both source and receiver, and three dimensional noise fields, the prediction model tends to be expensive in computer time and memory requirements. The ASONAR model presented here incorporates the RUMBLE bistatic reverberation model [H. P. Bucker, *J. Acoust. Soc. Am. Suppl.* 1 **80**, S 63 (1986)] and the DUNES directional noise model [R. W. Bannister, A. S. Burgess, and D. J. Kewley, *J. Acoust. Soc. Am. Suppl.* 1 **80**, S65 (1986)]. These newly developed models incorporate fast execution speeds and can run on IBM PC type microcomputers. By using SALT (Sound-Angle-Level-Time) tables, the propagation loss and reverberation to the many scattering areas are quickly determined. The scattering areas are increased in size as the ranges increase to keep the computation time within reasonable limits. The noise calculations are made using simplified propagation laws. ASONAR combines the outputs of RUMBLE and DUNES to calculate bistatic signal excess versus range from the receiver. Comparisons with other models and some experimental data are made. [Work supported by the NORDA NOP Program and NUSC.]

9:39

II17. The design and performance analysis of barrel stave projectors. D. F. Jones and G. W. McMahon (Defence Research Establishment Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

There is growing interest in low-frequency, light-weight sound sources for underwater acoustics research and for potential future sonar applications. A design being studied at DREA is the barrel stave projector, which consists of a piezoceramic cylinder driving a set of annular staves in flexure. An electroelastic finite element model has been used to analyze the linear vibrations and acoustic radiations of projectors having outer diameters of about 8 cm and fundamental resonance frequencies below 1600 Hz in water. Prototype projectors that operate at electroacoustic efficiencies greater than 50% at resonance have been assembled and their measured acoustic properties have been compared to those predicted by the model. In addition, a brief discussion of the model parameters and limitations is given.

9:43

II18. A practical VME-based sonar system development workstation. Shon Sloat (AP Labs, 4411 Morena Boulevard, Suite 150, San Diego, CA 92117)

This paper describes present capability and future development plans of a practical low-cost sonar system-development work station based on off-the-shelf VME equipment. The work station allows system engineers and algorithm designers to quickly develop sonar processing algorithms and systems. The work station can be used to (a) simulate and analyze the characteristics of real time environments for system and algorithm development, (b) concurrently develop final implementation software, and (c) generate analog test data for system test and operator instruction. The work station overcomes many of the difficulties experienced with developing sonar acoustic signal acquisition and processing systems: high design and maintenance costs of custom equipment; lack of controllable test data; inability to correlate simulation data with final system performance; and high software development costs, particularly with parallel, pipelined signal processing equipment. The sonar development work station hardware consists of a 16.67-MHz 68020 processor supported by a VME bus architecture. The hardware provides (a) interactive color graphics, (b) Sky Warrior and Mercury Zip 3232 array processors. The work station software is based on the Unix development system, VRTX real time kernel, X-Windows standard color graphics software, and array processor sonar processing primitives. This foundation provides software portability and device independent graphics.

Session JJ. Physical Acoustics V: Acoustic and Seismic Coupling I: Air-Solid Interfaces

Gilles A. Daigle, Chairman

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

Chairman's Introduction—9:00

Invited Papers

9:05

JJ1. Near-ground sound fields and surfaces of finite impedance. Tony F. W. Embleton and Gilles A. Daigle (Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

Studies over the past 30 years have elucidated the effects and interactions of many mechanisms involved in sound propagation outdoors. Interactions with the ground surface are especially important for many applications to practical problems. It is particularly significant that ground surfaces are porous, and thus have a finite complex acoustic impedance. The history of sound field measurements to deduce values of impedance, and its direct measurement, is reviewed. Such data prompted, and were then used to validate, models for ground impedance, first in terms of a single parameter (flow resistivity of the ground surface) and later in terms of one to three additional parameters. Impedance values for many grounds at low frequencies are difficult to measure accurately and several new techniques have been developed. Nonisotropic and nonhomogeneous ground has been modeled as a layered medium or as a fluid-filled porous matrix. These efforts have increased our understanding of the acoustical properties of the ground as an air-solid interface from a purely empirical basis to a detailed physical description based on the microscopic properties of the ground.

9:35

JJ2. Airborne sound to seismic coupling: Background for development of the models. James M. Sabatier, Henry E. Bass, and Lee N. Bolen (Physical Acoustics Research Laboratory, The University of Mississippi, University, MS 38677)

The initial measurements of the acoustic-to-seismic coupling phenomena using loud speakers as sound sources were made by the University of Mississippi and Waterways Experiment Station. For these measurements, a speaker was suspended from a crane, the sound level in the farfield was measured with a microphone above the surface, and the geophone response was measured below the surface. The results of this work largely indicated that the geophone had a response 1000 times greater than one would expect from calculations based upon simple acoustic transmission through a boundary between two perfect fluids. Another finding was that the transit times from speaker to microphone and speaker to geophone were approximately the speed of sound in the air. Success in our lab with water-saturated sediments suggested the application of the Biot model to the ground. Microphones were developed that were used as a pore fluid probe in the soil, and the attenuation and phase speed in the pore fluid were measured. This work led to the understanding of the commonly assumed locally reacting property of the ground.

10:05

JJ3. Airborne sound to seismic coupling: Present understanding. K. Attenborough (Engineering Mechanics Discipline, The Open University, Milton Keynes MK7 6AA, United Kingdom) and J. M. Sabatier (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677)

Theories of outdoor sound propagation at near-grazing incidence on porous ground surfaces assume either that the ground surface is an impedance boundary or that it is one of extended reaction. Such theories are consistent with models of the acoustic behavior of the ground as a locally reacting fluid or as a semi-infinite or layered modified fluid supporting a single type of compressional wave motion. However, these models are unable to explain the response of buried geophones to airborne sound sources. Measurements of the ratio of a shallow buried geophone output to the output of a vertically separated microphone placed near the ground surface have been explained in a variety of soil types by means of a layered poroelastic ground model. The layered poroelastic ground model and its experimental validation are described, and its implications for the impedance-frequency characteristics of outdoor ground surfaces are explored.

10:35

JJ4. Seismic, acoustic, and ionospheric wave kinematics associated with moderate to large earthquakes. S. I. Warshaw, F. E. Followill (Lawrence Livermore National Laboratory, Livermore, CA 94550), J. M. Mills, Jr. (SOHIO Production, Dallas, TX 75221), and P. R. Albee (Radio Sciences and Instrumentation, Santa Fe, NM 87501)

Unusually detailed Doppler radar records of ionospheric disturbances following two moderate-to-large dip-slip earthquakes were reported in 1984. These indicate that earthquakes can perturb the ionosphere by means of acoustic waves launched by the moving ground surface. For one earthquake [magnitude 6.5, Coalinga, CA, 2 May 1983 [J. H. Wolcott *et al.*, *J. Geophys. Res.* 89, 6835–6839 (1984)]]], a seismic-aeroacoustic-ionospheric kinematic wave analysis was performed to explain radar signature timing. Surface Rayleigh waves were modeled and interpolated from main and after-shock seismic records, acoustic wave trajectories by acoustic ray tracing in a standard atmosphere were established, and radar detection altitudes were determined from radio rays traced through an ionosphere calculated from an ionogram measured close by. Our calculated results agree well with Doppler signature times observed on several 5- and 10-MHz radar beam paths 160–300 km from the epicenter. Similarities and differences were identified in the other earthquake [magnitude 7.1, Urakawa-Oki, Japan, 21 March 1982 [T. Tanaka *et al.*, *J. Atm. Terr. Phys.* 46, 233–245 (1984)]] and its signatures were studied to assess commonalities in perturbation mechanisms. It is tentatively proposed that nonlinear acoustic propagation effects could be significant in shaping observed Doppler signatures. [Work performed under the auspices of USDOE by LLNL under contract W-7405-Eng-48.]

10:50

JJ5. Acoustic-to-seismic coupling under winter conditions. Lindamae Peck (USACRREL, 72 Lyme Road, Hanover, NH 03755-1290)

A field program to investigate acoustic-to-seismic coupling under winter conditions took place in Maine during December 1985–January 1986. The test area was packed sand with a top layer (30 cm) of loose sand. Site conditions were: 20-cm frost with no snow cover, 6-cm snow or 18-cm snow; and 45-cm frost, 28-cm snow. Microphones were at heights of 1/2–2 m or at the sand (snow) surface. Geophones were placed at the snow surface, the sand surface, and several depths in the sand layer. Acoustic-to-seismic coupling through frozen sand was investigated under controlled conditions using geophones in a sandbed at the USACRREL Frost Effects Research Facility. Test conditions were: dry sand, frozen or unfrozen; saturated sand, frozen or unfrozen; and thawing sand. The acoustic source for both investigations was blank pistol fire. Results on acoustic-to-seismic coupling through a snow layer and/or frozen sand are presented and contrasted with summer conditions for the frequency band 5–500 Hz. [Work supported by Directorate of Research and Development, U. S. Army Corps of Engineers Project 4A762730AT42.]

11:05

JJ6. Low-frequency acoustic-to-seismic coupling in the summer and winter. Donald G. Albert (USACRREL, 72 Lyme Road, Hanover, NH 03755-1290)

Experiments conducted in northern Vermont investigated acoustic-to-seismic coupling in the 5- to 500-Hz frequency band for ranges between 1 and 300 m. The strongest coupling into the ground occurs as the air wave passes, with a measured ratio of about $0.001 \text{ cm s}^{-1}/\text{Pa}$. The *P* waves are induced into the ground immediately under the source, and are the first

arrivals, since they travel at the higher seismic wave velocity, but their amplitudes are about three orders of magnitude less than the motion coupled via the later-arriving air wave. A comparison of the summer and the winter recordings shows two major effects of a 25-cm-thick snow cover. Frequencies above 100 Hz are strongly attenuated, with signals recorded by geophones above and below the snow yielding a value of around 2 for the *Q* of snow at 30 Hz; the snow produces a strong waveguide effect that enhances the air-coupled Rayleigh waves. [Work supported by Directorate of Research and Development, U. S. Army corps of Engineers Project 4A762730AT42.]

11:20

JJ7. Application of ray theory to propagation of low-frequency noise from wind turbines. James A. Hawkins and David T. Blackstock (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

Ray theory has been used to attempt to explain the propagation of very low-frequency (1–20 Hz) noise generated by downwind wind turbines. Two NASA field experiments with a large downwind machine at Medicine Bow, Wyoming, show that the downwind sound decays by spherical spreading near the source, but by cylindrical spreading downrange. Ray theory calculations explain this behavior. The favorable wind gradient downwind causes rays to refract downward and bounce along the ground. A sound channel is built up by repeated jumps in the number of ray arrivals at the receiver as downwind range is increased. Near the source, where no multiply reflected rays arrive, the sound field spreads spherically. The first jump (onset of multiple arrivals) is predicted to occur at about 2 km for the conditions of the NASA experiments. The second NASA experiment confirms this prediction. For the upwind sound, a shadow zone is predicted because of the unfavorable wind gradient. The NASA measurements show only that the upwind SPL in the predicted shadow zone is of order 7–15 dB less than the downwind SPL at comparable ranges. [Work supported by NASA.]

11:35

JJ8. Beam-to-mode conversion of a high-frequency Gaussian *P*-wave input in an elastic plate embedded in vacuum. I. T. Lu, L. B. Felsen, and J. M. Klosner (Polytechnic University, Farmingdale, NY 11735)

Gaussian beams, either individually or as synthesizing basis fields, provide useful models for the response due to Gaussian or more generally shaped high-frequency source inputs, respectively. When propagating into a layered environment, an initially well-collimated beam undergoes diffusion after successive reflections, and is converted essentially into the oscillatory pattern of one or more guided modes. The diffusion process is accelerated in the presence of multiwave coupling at interfaces, for example, the *P*-*SV* coupling at the free boundary of an elastic solid. This phenomenon is examined here for a two-dimensional *P*-wave beam input into an elastic plate embedded in vacuum. The beam is modeled via the complex source point method whereby a Gaussian-waisted input is generated from a line forcing function by assigning complex values to the source coordinates. This scheme avoids the need for plane-wave spectral decomposition and synthesis, but the analytic continuation to complex source coordinates of various line force Green's function representations in a layered medium requires precautions, which are discussed. Numerical calculations of *P* and *SV* potentials, stresses and motions, performed by beam tracking and mode summation at successive cross sections, reveal the beam-to-mode transition process, and suggest that a hybrid beam-mode algorithm provides the most direct and cogent physical explanation. [Work supported by AFOSR and ONR.]

11:50

JJ9. Acoustic wave interactions at a mean-flow stagnation point. Charles Thompson and Martin Manley (Department of Electrical Engineering, University of Lowell, One University Avenue, Lowell, MA 01854)

The interaction between acoustic disturbances and the mean-flow

near the stagnation point of a bluff body will be examined. The stability of such flows will be investigated. It will be shown that streamwise vorticity generated by the Stokes layer can enhance the receptivity of the mean-flow boundary layer to free-stream disturbances. The downstream evolution of the vortical motion of the fluid will also be modeled and the results will be presented. [Work supported by Analog Devices Professorship.]

THURSDAY MORNING, 19 NOVEMBER 1987

TUTTLE SOUTH ROOM, 9:00 TO 11:35 A.M.

Session KK. Physical Acoustics VI: General Topics in Physical Acoustics

Michael E. Haran, Chairman

IBM Federal Systems Division, Manassas, Virginia 22110

Chairman's Introduction—9:00

Contributed Papers

9:05

KK1. Measurement of the universal gas constant R using a spherical acoustic resonator. M. R. Moldover, J. P. M. Trusler, T. J. Edwards, J. B. Mehl, and R. S. Davis (Thermophysics Division, Center for Chemical Engineering, National Bureau of Standards, Gaithersburg, MD 20899)

A spherical acoustic resonator has been used to redetermine the universal gas constant R with an uncertainty of 1.8 ppm (standard deviation). To accomplish this, three subtasks were completed. (1) The volume of a spherical shell was determined by weighing the mercury required to exactly fill it at the temperature of the triple point of water, 273.16 K. (2) With the resonator filled with commercially supplied argon, the resonance frequencies of the radial modes were measured as a function of pressure. Using our theoretical model for the cavity, the frequency measurements were combined with the mean resonator radius determined in subtask (1) to obtain the speed of sound in commercially supplied argon. (3) Finally, the speed of sound in the commercially supplied argon was compared to the speed of sound in a "standard" sample whose chemical and isotopic composition was accurately established.

9:20

KK2. Effects of carbon dioxide and humidity on some acoustic and thermodynamic properties of air. George S. K. Wong (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

The theoretical data on the combined effects of humidity and carbon dioxide content on some physical properties of air, such as the characteristic impedance ρc and the sound speed c , the specific heats C_p and C_v , and their ratio γ , and the density ρ , have been studied. In general, over the temperature range 0°–30 °C, the normalized values $C_p/(C_p)_0$, $C_v/(C_v)_0$, and c/c_0 , become larger with the inclusion of humidity and rising temperature; and they are also inversely proportional to the CO₂ content. Similarly, ρ/ρ_0 and $\rho c/(\rho c)_0$ are proportional to the CO₂ content, but they are inversely proportional to humidity and temperature. However, the normalized value γ/γ_0 becomes smaller with the increase of humidity, temperature, and CO₂ content. The above reference values, which are indicated with a zero suffix, such as $(C_p)_0$ and ρ_0 , refer to dry standard air (314 ppm CO₂ content) at 0 °C and at a pressure of 101.325 kPa.

9:35

KK3. Eigenmodes of quasicrystals. J. D. Maynard and Shanjin He (Department of Physics, The Pennsylvania State University, University Park, PA 16802)

Recently, a new state of matter, referred to as quasicrystalline, was

discovered. Previously, solids could be classified as crystalline, with periodic lattice spacing, or as glassy, with random site spacing. The new quasicrystal structures appear to have long-range order, showing sharp peaks in the Fourier transform space as in a periodic system, but they also have properties that are impossible for any periodic structure, such as fivefold rotational symmetry. For periodic systems, Bloch's theorem may be used to understand physical properties such as wave transmission, and it is of current interest to learn if any such symmetry theorems apply to quasiperiodic systems. Rigorous theorems for one-dimensional quasiperiodic patterns based on a Fibonacci sequence suggest that such a pattern may be useful in control of vibration transmission in rib-stiffened plates. However, in two and higher dimensions, little is known about the consequences of quasiperiodic structure. Recently, acoustic measurements have been made on a two-dimensional quasiperiodic system and the frequency spectrum, density of states, and eigenmode patterns, showing some features unique to the quasiperiodic pattern, have been determined. [Work supported, in part, by NSF DMR 8701682 and the Office of Naval Research.]

9:50

KK4. Interaction of a sound beam with a two-fluid interface. Jacqueline Naze Tjøtta (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway, and Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029), Hanne Sagen (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway), and Sigve Tjøtta (Department of Mathematics, The University of Bergen, 5007 Bergen, Norway, and Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

The reflection and transmission of a real sound beam at the interface between two homogeneous and dissipative fluid layers are considered. Numerical results are obtained by using a fast Fourier transform algorithm. For the transmitted field, they show that, at a given incident angle, the direction and displacement of the beam depend critically on the absorption coefficient, and on the distance between the source and the interface. Various asymptotic formulas are also presented, which allow for a physical interpretation of the numerical results.

10:05

KK5. The role of cutoff modes in waveguides with boundary discontinuities. R. Sen (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24060) and Charles Thompson (Department of Electrical Engineering, The University of Lowell, Lowell, MA 01854)

When a waveguide with a boundary discontinuity is excited by a low-frequency plane wave, cutoff cross modes are induced by the discontinuity.

uity. Although these modes are discernible only in the vicinity of the discontinuity, they act as sources in the incompressible local flow, and thereby affect the mean acoustic pressure. This has a profound effect on the farfield reflection and transmission coefficients R and T , and, consequently, on the junction impedance Z . The method of matched asymptotic expansions shall be used and it will be shown that, by properly accounting for the cross modes, the frequency range of validity of asymptotic estimates of R , T , and Z can be greatly extended. The possibility of improving these estimates by the use of Padé fractions will also be discussed. Finally, the formal difficulties of incorporating cutoff modes into the acoustic field description are discussed, and an extension of the method of matched asymptotic expansions to circumvent them is presented.

10:20

KK6. Hot-film anemometer response to hydroacoustic particle motion under imposed bias flow. Pieter S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

The response of a hot-film anemometer to vertical hydroacoustic particle motion is influenced by free convection, which acts as a bias flow. The output was shown to be proportional to particle displacement for a wide range of parameters [P. S. Dubbelday, *J. Acoust. Soc. Am.* **79**, 2060–2066 (1986)]. It was expected that an imposed bias flow would increase the output and remove the dependence on the direction of gravity. Therefore, a hot-film sensor (diameter d) was subjected to an underwater jet from a nozzle. The ratio of the voltage outputs, due to a hydrostatic field, with and without this dc flow, was measured as a function of frequency for various flow speeds. The output showed a transition from being proportional to particle speed, to being proportional to particle displacement, depending on the angular frequency ω and imposed flow speed v . The transition takes place when a dimensionless number Ω , defined as $\Omega = \omega d / v$, is of order 1. This phenomenon has consequences for the interpretation of the measurement of turbulence, where it is customary to use the dc velocity calibration without reference to frequency dependence. [Work supported by the ONR.]

10:35

KK7. On the extension of the acoustic variational principle, from media at rest to potential mean flows. L. M. B. C. Campos (Instituto Superior Técnico, 1096 Lisboa Codex, Portugal)

The well-known acoustic variational principle for sound in a medium at rest [e.g., Morse and Ingard, *Theoretical Acoustics* (McGraw-Hill, New York, 1968), p. 248], is generalized to the acoustics of a potential mean flow, and is used to derive: (i) the energy equation for sound in free and ducted flows; (ii) the wave equation in an inhomogeneous fluid at rest, in low Mach number and in high-speed convection. (iii) the Websters' horn wave equations and wave equations for low Mach number and high-speed nozzles. Case (iii) above applies to quasi-one-dimensional sound propagation in nonuniform ducts, and generalizes the variational principle of Weibel [J. Acoust. Soc. Am. **75**, 105–106 (1955)] to include the effects of potential mean flow. The linear acoustic wave equations derived from the variational principle are checked by comparison with exact nonlinear equations obtained by a distinct method.

10:50

KK8. Three-dimensional analysis for radial and thickness mode excitation of piezoceramic disk. Michel Brissaud (Electrical

Engineering and Ferroelectrics Laboratory, INSA, 69621 Villeurbanne Cédex, France)

Standards on piezoelectricity [IEEE Trans. Sonics Ultrason. **SU-31** (March 1984)] describe the analysis of vibrations on piezoelectric materials having simple geometrical shapes. The results are based on linear piezoelectricity, and resonance modes are treated as noncoupled vibrational modes. However, in general, real materials involve coupled modes. In addition, they may have one dimension that is not greater than the others. Therefore, it is not possible to state concisely a specific set of conditions under which the definitions and equations contained in the standards apply. However, for pure-mode direction propagation, the mechanical displacements are only related to a single coordinate, and the electrical impedance and general electromechanical coupling factor equations can be determined. With the same assumptions as in the thickness excitation of plates [M. Brissaud, *J. Acoust. Soc. Am. Suppl.* **1** **79**, S46 (1986)], that is, completely stress-free boundary conditions and lossless piezoelectric materials, the impedance equation of the disk can be given. It should be noted that, when the piezoelectric coefficients e_{13}^D and h_{33} vanish, the impedance reduces to an equation that is similar to that given by Mason's one-dimensional model [Phys. Acoust. **1** (1964)].

11:05

KK9. Photoacoustic effect from fine particles: Grain size and frequency dependence of the acoustic yield. J. Pelz, U. Netzelmann (Institut für Experimentalphysik, Ruhr Universität, D-4630 Bochum 1, Federal Republic of Germany), and D. Schmalbein (Bruker Analytische Messtechnik GmbH, D-7512 Rheinstetten, Federal Republic of Germany)

The audioacoustic response of air in contact with fine copper-sulfate particles that are heated periodically by resonant microwave absorption has been investigated as a function of the modulation frequency (10 Hz to 5 kHz) of the microwave power for different particle sizes (10 μ m to 1 mm). Absolute values of the photoacoustic (PA) yield are obtained by normalization with the simultaneously detected conventional EPR signal. The PA-signal generation efficiency has been found to increase strongly with decreasing particle size with the maximum value occurring at a distinct size-dependent frequency. The size and frequency behaviors of the amplitude and phase shift of the PA-signal are analyzed theoretically on the basis of a model that considers a single isolated particle.

11:20

KK10. Extrapolation of acoustic fields by means of source function estimation. G. Elias and F. Payen (Department of Physics, Office National d'Etudes et de Recherches Aéropatiales, BP 72, 92322 Châtillon Cedex, France)

Extrapolation of acoustic pressures is needed when farfield measurements are not possible. Most of the methods usually used are based on a direct application of Kirchhoff's integral or on spectral decomposition, which leads to a fine sampling of a closed surface. The present method requires only a limited number of microphones and consists of a source function estimation. In the first step, the microphone array is used as a focused antenna for measuring the overall extent of the source region. In the second step, a biorthogonal basis is computed for the expansion of any source function limited to the previous spatial extent and the pressure radiated on the array. Then, by projecting the measured pressure on this basis, an estimate of the source function can be obtained to extrapolate the acoustic pressure in the farfield. Simulations show a good agreement between real and estimated directivity diagrams and the precise limitations of this method. [Work supported by DCN.]

Session LL. Speech Communication V: Perception (Poster Session)

Linda Polka, Chairman

*Department of Communicology, CBA 229, University of South Florida, Tampa, Florida 33612***Contributed Papers**

All posters will be displayed from 9:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 to 10:30 a.m., and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

LL1. Individual differences in the perception of cues for initial stop place and voicing contrasts. Valerie Hazan (Department of Phonetics and Linguistics, University College London, 4, Stephenson Way, London NW1 9DY, United Kingdom)

Fifteen normal-hearing adults were tested on their ability to identify synthesized stimuli from three initial stop contrasts. Two place contrasts, BAIT-DATE and DATE-GATE, were presented in three conditions. In the "full-cue" condition, the place contrast was signaled by changes in burst frequency and $F2/F3$ transition. In the two "reduced-cue" conditions, the contrasts were signaled by only one of these two cues. A voicing contrast, GATE-KATE, was presented in two conditions: a "full-cue" condition, in which the contrast was marked by changes in VOT and $F1$ onset; and a "VOT cue" condition in which the $F1$ onset was flat. Group results show significant variance across conditions, both in terms of the phoneme boundaries and identification function gradients obtained. In addition, individual differences were found in the labeling of reduced-cue conditions. Some subjects showed little difference between the labeling of the full-cue and the reduced-cue conditions of the BAIT-DATE and GATE-KATE contrasts, while others were totally unable to establish a contrast for some conditions.

LL2. Perception of S-to-unvoiced-stop transitions in natural and synthetic speech. Jan P. H. van Santen and Beverly M. Glasgow (AT&T Bell Laboratories, Murray Hill, NJ 07974)

The perceptual role of the preclosure portion of S-K, S-T, and S-K transitions was investigated. Mahalanobis distances between reflection coefficient vectors (derived from linear predictive coding analysis) showed that differences between the stops are as large before the closure as during the release. In a paired-comparison experiment using natural speech, listeners preferred consistent words (e.g., S-T followed by TOUT) over inconsistent words (e.g., S-K followed by TOUT). However, perceptual differences were subtle, and the stop identity was never in question. In a subjective quality rating experiment using synthetic speech, there were no differences between consistent words and stop-neutral words (S-silence followed by, e.g., TOUT), but inconsistent words were given lower ratings. It is concluded that the perceptual role of the preclosure portion of S-to-unvoiced stop transitions is small, and can be ignored in current speech synthesizers. This apparent conflict between the experimental results and the Mahalanobis distance analysis may involve the perceptual saliency of the dimensions along which vectors differ and the effects of amplitude. Finally, the results demonstrate the sensitivity of subjective quality ratings.

LL3. Recognition of consonant clusters. Moshe Yuchtman (Project Phoenix of Madison, Inc., 2001 S. Stoughton Road, Madison, WI 53716), Joseph Stemberger (Department of Linguistics, University of Minnesota, Minneapolis, MN 55455), and Christopher Martin (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Perceptual processing of speech has been frequently studied using analyses of consonant recognition errors. Typically, nonsense CV or VC

syllables are used as stimulus materials. Nonsense consonant clusters constitute a class of sounds that are largely devoid of semantic context, yet they represent a higher level of phonological organization. Consequently, the recognition of these sounds may reflect processing levels that are not involved in the perception of simpler stimuli. In the present study, consonant clusters (CCV or CCCV in /a/ vowel context), as well as their singleton members, were presented to listeners against a background noise at 10, 5, and 0 dB S/N. The results show systematic trends in terms of the frequency and the patterns of recognition errors. In general, the proportion of substitutions, additions, and deletions resemble the pattern of errors observed in production errors of consonant clusters. These and further results will be presented and discussed at the meeting.

LL4. Auditory factors in the perception of stops and glides. Margaret A. Walsh and Randy L. Diehl (Department of Psychology, University of Texas, Austin, TX 78712)

There is some disagreement in the literature about the relative contribution of formant transition duration and amplitude rise time in cuing the stop-glide distinction (e.g., [b] vs [w]). Lack of resolution on this point is partly due to the confounding of these two variables in most studies. For a series of identification experiments, sets of synthetic [ba] and [wa] stimuli were created in which transition duration and rise time varied orthogonally. Both variables affected labeling performance in the expected direction, but transition duration was by far the more important factor. A similar pattern of results was obtained for single sine-wave stimuli that modeled the rise times, frequency trajectories, and durations of the first formant in the [ba] and [wa] stimuli. A likely auditory basis for both the speech and nonspeech effects is that rapid frequency transitions and short rise times both contribute to a higher degree of spectral splatter (i.e., dispersion of energy across a wide frequency band), which may be a principal cue for abrupt onsets in general. [Work supported by NICHD.]

LL5. Identification of stop consonants from continuous speech in limited context. Lori F. Lamel (Room 36-545, Department of Electrical Engineering and Computer Science, and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

This paper describes a series of perceptual experiments aimed at evaluating listeners' ability to identify singleton stop consonants. The stimuli consisted of portions of speech extracted from a corpus of about 3600 sentences spoken by over 450 talkers. A variety of factors, such as the stress pattern, syllable position, and phonemic environment, were systematically varied to determine the extent to which these factors influence the listeners' decision. At least 20 listeners heard syllable-initial stops in one task and nonsyllable-initial stops in another. In each task, only the immediate surrounding context was available. Overall identification rates were better than 97% for syllable-initial stops and 85% for the noninitial stops. In the syllable-initial case, 75% of the errors were in voicing, predominantly voiced stops being heard as their voiceless counterparts. For the noninitial stops, 88% of the errors were also in voicing. However, in this task, most of the errors were voiceless stops heard as voiced. Almost 70% of the errors for the noninitial stops involved alveolars, whereas no partic-

ular place of articulation accounted for most of the errors for the initial stops. Factors other than syllable position will also be discussed, and comparisons will be made to other stop identification studies. [Work supported by DARPA under contract N00014-82-K-0727, monitored through the Office of Naval Research.]

LL6. The role of rate of transition in place perception. Renee A. E. Zakia and John Kingston (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

Kewley-Port, Pisoni, and Studdert-Kennedy ["Perception of static and dynamic acoustic cues to place of articulation in initial stop consonants," *J. Acoust. Soc. Am.* 73, 1779-1793 (1983)] showed that initial bilabial and alveolar stops can be reliably identified from information in the first 20 ms of the syllable, including the burst and the beginning of the transition. Identification of stops as velars required a longer stimulus, indicating that a slow change in formant frequencies is characteristic of this place of articulation. If rate of change of formant frequencies is a positive cue to the identification of velar stops, it would be predicted that varying rate of change would affect their identification. To test this prediction, synthetic stop-vowel stimuli, derived from natural tokens, consisting of all combinations of [b,d,g] and the vowels [i,a], were edited to produce stimuli with four rates of formant transition: 100, 75, 50, and 25 ms. Listeners in this experiment identified place of articulation consistently regardless of rate of transition; specifically, there was no significant interaction between place and rate of transition. This would indicate that a slow formant transition is not necessary if a stop is to be identified as a velar. There was, however, a significant interaction between vowel and place of articulation, which implies that the magnitude, if not the rate, of the transition is a significant cue to place.

LL7. Rate versus duration of formant transition as a discriminative cue. Robert J. Porter, Jr. (Kresge Hearing Research Laboratory, Department of Otorhinolaryngology, LSU Medical Center, New Orleans, LA 70112 and Department of Psychology, University of New Orleans, New Orleans, LA 70122), Jane Collins, John K. Cullen, Jr. (Kresge Hearing Research Laboratory, Department of Otorhinolaryngology, LSU Medical Center, New Orleans, LA 70112 and Department of Communication Disorders, College of Arts and Sciences, Louisiana State University, Baton Rouge, LA 70804), and Dena F. Jackson (Kresge Hearing Research Laboratory, Department of Otorhinolaryngology, LSU Medical Center, New Orleans, LA 70112)

The linguistic messages of speech cannot be clearly read in its time-by-frequency-by-amplitude signature nor is it manifest in the corresponding psychoacoustic domain. It is suspected that important aspects of speech messages may, instead, be revealed in higher-order, rate-of-change attributes of speech signals and their corresponding perceptual representation. In a recent exploration of rate-of-change-of-spectrum [Besing *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 81, S54 (1987)], 300-ms, second-formant-like signals were used, which were varied in onset frequency to produce variations in the rate of a 30-ms, initial transition. Results suggested listeners may use rate-of-frequency-change to discriminate among these signals. The present study examined thresholds for rate changes produced by transition *duration* variation instead of onset frequency. Transitions began at either 1500 or 2100 Hz and rose or fell to a steady state of 1800 Hz. Standard transition durations of 60 and 124 ms were used. Discrimination was compared for rising and falling transitions presented either alone, or presented together with first-formantlike resonances. Results for ten subjects suggest, as in the previous study, that listeners may use perceived rate change rather than perceived duration to discriminate these speechlike signals. [Work supported by NIH-NINCDS and Louisiana Lions Foundation.]

LL8. Perceptuomotor adaptation to natural speech adaptors. Linda I. Shuster (Department of Speech Pathology and Audiology, West Virginia University, Morgantown, WV 26506-6122)

Perceptuomotor adaptation has been demonstrated using synthetic adaptors [Cooper and Nager, *J. Acoust. Soc. Am.* 58, 256-265 (1975); Jamieson and Cheesman, *J. Phonet.* 15, 15-27 (1987)]. However, similar

results have not been obtained when natural adaptors were employed [Summerfield *et al.*, *J. Phonet.* 8, 491-499 (1980); Shuster and Fox, *ASHA* 28, 72 (1986)]. In the present study, subjects produced examples of the token [rə^hi] prior to adaptation and then after perceptual adaptation with a naturally produced [rə^hi]. Voice onset time was then measured for each spoken utterance. It was found that VOTs were significantly shorter after adaptation to the bisyllable than they were prior to adaptation. These results are similar to those obtained with synthetic adaptors and support the notion that perceptuomotor adaptation reflects a perception-production link.

LL9. Central and peripheral representation of whispered and voiced speech. Arthur G. Samuel (Department of Psychology, Box 11A Yale Station, New Haven, CT 06520)

Whispered speech is very different acoustically from normally voiced speech yet listeners have little trouble perceiving whispered speech. Two selective adaptation experiments explored the basis for the common perception of whispered and voiced speech, using two /ba/-/wa/ continua (one voiced, one whispered). The first experiment used the endpoints of each series as adaptors, and several nonspeech adaptors as well. Speech adaptors produced reliable labeling shifts of syllables matching in periodicity (i.e., whispered-whispered or voiced-voiced); somewhat smaller effects were found with mismatched periodicity. A periodic nonspeech tone with short rise time produced adaptation effects like those for /ba/. These shifts occurred for whispered test syllables as well as voiced, indicating a common abstract level of representation for voiced and whispered stimuli. Experiment 2 replicated and extended experiment 1, using same-ear and cross-ear adaptation conditions. There was perfect cross-ear transfer of the nonspeech adaptation effect, again implicating an abstract level of representation. The results support the existence of two levels of processing for complex acoustic signals. The commonality of whispered and voiced speech arises at the second, abstract level. Both this level, and the earlier, more directly acoustic level, are susceptible to adaptation effects. [Work supported by AFOSR.]

LL10. Modeling the perception of vowel-like formant peaks. Hector Javkin, Brian Hanson, Paul Neyrinck, and Hisashi Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

An improved model of the relationship of harmonic structure to vowel perception is proposed and evaluated. Javkin, Hermansky, and Wakita [11th International Congress of Phonetic Sciences, Tallinn, Estonia, USSR (1987)] compared listeners' responses to one-formant stimuli with the computed weighted average of the two most prominent harmonics [the most important frequency, or MIF of Carlson, Fant, and Ganstrom, *Auditory Analysis and Perception of Speech* (Academic, London, 1975)] applied with different scaling factors. With this measure, a relatively expanded scale such as magnitude comes closest to perceptual test results. To improve on the characteristics of MIF, a critical band analysis was adopted and Chistovich and Chernova [Speech Commun. 5, 3-16 (1986)] were followed in applying a center of gravity formant estimation (CG). Because CG is highly dependent on the integration interval, an iterative process was used, with each iteration choosing the frequency limits for the next analysis until convergence. Initial results suggest that, while CG is less affected by amplitude expansions than MIF, the magnitude space most closely approximates listeners' responses for the one-formant stimuli. Evaluations of the method with multiformant stimuli are also presented.

LL11. Acoustic versus phonological context effects in the perception of formant transition continua. Terrance M. Nearey and Sherrice E. Shammass (Department of Linguistics, University of Alberta, Edmonton T6G 1E7, Canada)

The effect of the following vowel on listener's judgments of place of articulation in formant transition continua is perhaps the archetypal example of context sensitivity in speech perception. Is this sensitivity explicable in terms of acoustic context alone or must the judged phonological

identity of the following vowel also be considered? Shamma [unpublished dissertation, University of Alberta (1985)] designed transition-continua experiments involving ambiguous and nonambiguous vowel stimuli to examine this question in detail. Linear logistic analysis of these data confirms the systematic role of acoustic context effects, but indicates that the phonological context effects are of a highly restricted nature and, furthermore, are both task and listener dependent. A summary of this analysis is presented and the results are discussed in the context of other results from the literature (including those of Mermelstein, Massaro, Ohala, and Repp). The evidence is consistent with the hypothesis that phoneme-sized units (rather than diphones or larger units) serve as the primary interface between acoustic cues and phonological units.

LL12. Identification of "hybrid" vowels in sentence context. James J. Jenkins and Winifred Strange (Department of Psychology, University of South Florida, Tampa, FL 33620)

It has been demonstrated that listeners can identify the intended vowel in a CVC syllable even when the vowel nucleus has been attenuated to silence, leaving only onglides and offglides (silent center syllables). Verbrugge and Rakerd [Lang. Speech 29, 39-57 (1986)] constructed "hybrid" silent center syllables by cross splicing onglides and offglides of citation-form CVC syllables spoken by a male and a female talker such that formant trajectories were discontinuous. Identification of the intended vowel in hybrid syllables was no less accurate than vowel identification of the single-talker silent center syllables. The present study replicates this research with syllables spoken in a carrier sentence, "I say the word /dVd/ some more," using ten American-English vowels. Hybrid silent center syllables were prepared by crossing the stimulus sentences so that the sentence started with one talker and switched to the other after the silent portion of the test syllable. Silent center stimuli were prepared for each talker separately as controls. Vowel identification was as accurate in the hybrid condition as in the control conditions suggesting that dynamic syllable structure provides talker-independent information about the articulatory/acoustic event. [Research supported by NINCDS.]

LL13. Classification of vowel productions by means of perceptual target zones: A response to Ladefoged and Studdert-Kennedy. James D. Miller (Central Institute for the Deaf, Saint Louis, MO 63110)

In discussion of my paper [Miller, J. Acoust. Soc. Am. Suppl. 81, S16 (1987)] at the 113th Meeting, Dr. Ladefoged and Dr. Studdert-Kennedy requested specific numbers regarding percent correct classifications of vowel productions achievable through use of the auditory-perceptual theory and its perceptual target zones (PTZs). At the time of the 113th Meeting, the estimated shapes and locations of the PTZs were very preliminary and represented only a second iteration (I2). The percents correct for the I2 zones were moderate (50%-95%). Based on reconsideration of older data and on numerous new measurements made in our laboratory, a new set of PTZs is being developed for the vowels. These represent the third iteration or I3 zones. The I3 zones will be presented and percentages of correct classifications for several data sets will be reported. Issues involved in developing decisive tests of the perceptual target zones will be discussed. [Supported by NINCDS and AFOSR.]

LL14. Auditory-perceptual approach to the stop consonants. James D. Miller and Allard Jongman (Central Institute for the Deaf, Saint Louis, MO 63110)

Using natural tokens and the synthetic stimuli of Abramson and Lisker [6th Internat. Congr. Phonet. (1970)], the auditory-perceptual theory shall be applied to syllable-initial stop consonants. The burst-friction components of each token are analyzed for the locations of their second and third sensory formants, BF2 and BF3. These are located in the auditory-perceptual space (APS) by the formulas: $x = \log(\text{BF3}/\text{BF2})$ and $y = \log(\text{BF2}/\text{SR})$, where SR is the sensory reference. The glottal-source components are then analyzed for their sensory formants: SF1, SF2, and SF3. These are then located in APS by equations: $x = \log(\text{SF3}/\text{SF2})$, $z = \log(\text{SF1}/\text{SR})$, and $z = \log(\text{SF2}/\text{SF1})$. In this way, a sensory path comprised of burst-friction points and glottal-source points is gener-

ated. Next, a sensory-perceptual transformation is applied to generate a unitary perceptual path. Since the sensory-perceptual transformation is reactive, the perceptual path overshoots the burst-friction onset of the token and enters a physically unrealizable octant of APS, wherein the perceptual target zones for the stops are located. Distinct target zones are estimated for the stops [p,t,k,d,b,g], for h-like aspiration, and for voice bars. [Supported by AFOSR and NINCDS.]

LL15. Locations of the burst onsets of stops in the auditory-perceptual space. Allard Jongman (Central Institute for the Deaf, Saint Louis, MO 63110)

Preliminary LPC analyses of the voiceless stop consonants [p,t,k] in the context of the vowels [i,e,a,u,u] produced in real CVC words by one male and one female speaker suggest a method for locating formants in burst-friction spectra. Specifically, a 24-ms Hamming window was centered over the onset of the burst. First, the spectral peak with maximum amplitude below 6 kHz was located. Then, moving from left to right, the center frequencies of the first two peaks within 10 dB of the maximum were picked as the burst-friction components BF2 and BF3. Furthermore, in those cases where BF2 had been picked and BF3 was separated from BF2 by 2500 Hz or more, the frequency value for BF2 was also used for BF3. These burst-friction prominences were then converted into coordinates of the auditory-perceptual space: $x = \log(\text{BF3}/\text{BF2})$ and $z = \log(\text{BF2}/\text{SR})$, where SR is the sensory reference. This method of analysis enabled us to distinguish the stops, in terms of place of articulation, with good accuracy. [Work supported by AFOSR.]

LL16. Signal detection analyses of speaker identification accuracy. George Papcun (Los Alamos National Laboratory, Los Alamos, NM 87545), Jody Kreiman (Phonetics Laboratory, UCLA, 405 Hilgard, Los Angeles, CA 90024), and Anthony Davis (Department of Linguistics, Stanford University, Palo Alto, CA 94305)

The accuracy of speaker identification can be measured under various experimental paradigms including: (a) multiple-choice, closed-set experiments, in which listeners choose one speaker from among a lineup of n speakers [as in F. McGehee, "The reliability of the identification of the human voice," J. Gen. Psychol. 17, 249-271 (1937)]; (b) multiple-choice, open-set experiments, in which listeners choose one speaker from among a lineup of n speakers, but can also refuse to choose any of the speakers [as in C. P. Thompson, "Voice identification: Speaker identifiability and a correction of the record regarding sex effects," Human Learn. 4, 19-27 (1985)]; and (c) independent-judgment, open-set experiments, in which subjects are to make each choice independently of others [as in G. Papcun, J. Kreiman, and A. Davis, "Memory for unfamiliar voices at delays of one, two and four weeks," submitted for publication]. In this paper, it was demonstrated that signal detection analysis, with certain extensions, makes it possible to compare results across these diverse experimental paradigms, and to see the essential identity of apparently disparate results. A computer program is provided to treat case (b).

LL17. Long-term changes in voice characteristics: Implications for speaker identity verification (SIV). Timothy C. Feustel and George A. Velius (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

In a typical speaker identity verification (SIV) scenario, a talker first asserts an identity claim via some means other than speech. A reference sample of the claimed individual's speech is then loaded into a pattern matcher and compared against a new sample of the same word or phrase solicited on-line. Based on the similarity of the two utterances, and a predetermined criterion, a decision is made as to the veracity of the claimed identity. One difficult problem for SIV applications is that the spectral characteristics of a speaker's voice change over time in ways that are poorly understood. Utterances were collected from three speakers at regular intervals over a period of 23 weeks, and were analyzed first with respect to the magnitude of spectral dissimilarity over varying time intervals, and, second, with respect to the effect of these differences on error

rates for an SIV decision task. The data show an increase in mean utterance dissimilarity of about 8% over a period of 2 weeks, followed by a much more gradual increase over 6 months. While the absolute change in mean dissimilarity was small, it resulted in a doubling of the error rate for the SIV decision task from about 5% to 10%.

LL18. Aural signal processing in the context of a digital hearing model. D. M. Chabries, M. W. Christiansen, R. W. Christiansen, R. H. Brey, R. W. Harris (Brigham Young University, Provo, UT 84604), and M. Robinette (Mayo Clinic, Rochester, MN 55905)

A nonlinear model of the human aural system is proposed. Utilizing a homomorphic transformation, the model is implemented in a digital computer and accounts for the Fletcher critical bands in hearing, the nonlinear relationship between input sound intensity and perceived loudness known as recruitment, as well as the thresholds of audibility and the maximum tolerance threshold. The transformed or output signal domain is identified as a perceptual space. It is shown that the model can be used to reproduce the Fletcher Munson equal loudness contours for normal hearing populations. Modification of easily measured parameters results in the generation of equal loudness contours for hearing impaired individuals. Through a combination of the forward homomorphic transformation for normal hearing to the perceptual domain and the inverse mapping utilizing parameters of a hearing impaired individual, it is shown that hearing intelligibility is dramatically improved. It is postulated that signal processing performed in the perceptual domain will produce results more pleasing and intelligible to the listener. Results of auditory testing utilizing the proposed model for both normal and hearing impaired populations are presented. Speech intelligibility scores are shown to increase by more than 40% (i.e., from 40% to over 80%) over preprocessed scores under headphones for hearing impaired subjects used in these tests.

LL19. A model for the transduction stage of auditory speech processing. Stephanie Seneff (Room 36-549, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

A new model for the transduction stage of auditory processing is proposed. The model is applied independently to each of 40 critical band filter outputs, spanning the frequency range from 100–6400 Hz. The model consists of four components in cascade: an instantaneous nonlinear half-wave rectifier, a short-term adaptation component, a low-pass filter, and, finally, a rapid adaptation component. Model outputs were compared with available auditory data in a number of different dimensions as follows: (1) onset response to tone bursts, (2) forward masking effects, (3) steady-state behavior for vowel-like stimuli, (4) incremental response characteristics, and (5) synchrony loss at high frequencies. Two distinct spectral representations are computed from the model outputs: an amplitude spectrum corresponding to mean-rate response and a synchrony spectrum which makes use of temporal information to enhance spectral peaks. The model is currently being incorporated into a speaker-independent continuous speech recognition system under development. [Work supported by DARPA under Contract N00039-85-C-0254, monitored through Naval Electronics Systems Command.]

LL20. Pulse synchronous analysis of speech using an auditory representation. James R. Glass and Stephanie Seneff (Room 36-585, Department of Electrical Engineering and Computer Science, and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

A procedure has been developed that automatically extracts pulse excitation information from the speech signal. This time-domain algorithm operates on the transduction stage outputs of an auditory model developed by Seneff [this meeting]. Specifically, robust inflection points are determined from a waveform obtained by summing the channel outputs. This pulse-like representation provides information about periodic and aperiodic excitation, and can be used to extract periodicities in voiced regions. Thus the information provided by this signal complements the spectral information provided by mean-rate response outputs of the auditory model. Furthermore, this signal can be used to sample channel outputs pulse synchronously. This produces a spectral representation with

sharper temporal onsets and offsets, and with an enhanced dynamic range in the frequency dimension. [Work supported by DARPA under Contract N00039-85-C-0254, monitored through Naval Electronics Systems Command.]

LL21. Alternate measures of response bias in speech recognition by older listeners. Robin T. K. Zeller (Speech and Communicative Disorders, Adelphi University, Garden City, NY 11530) and Louis J. Gerstman (Department of Psychology, City College, CUNY, New York, NY 10031)

Fifty-two listeners ranging in age between 50 and 81 years, listened to two R-SPIN Tests, responding to each stimulus verbally and with a numerical confidence rating. Twelve performance measures were created: for HP and LP items separately, percent correct, d' and $P(A)$, which both measure sensitivity, and B , which measures response bias; for all 100 items, a mean and standard deviation of the ratings; and for the errors on the LP items, a mean frequency of word usage, and a rating of contextual feasibility by a panel of 15 judges. Correlations with age revealed no effects for the HP items, but established that with advancing age, there was lower sensitivity on the LP items as measured by d' and $p(A)$, but no change in B . Additionally, the older listeners used a narrower range of the rating scale and made more contextually feasible errors. No age variation was found for word frequency of errors, but all the proposed responses were ten times more probable than the misheard LP words.

LL22. Perceptual and acoustic effects of nonlinear signal processing. Moshe Yuchtman, Rick Jenison, and Robert Lasky (Project Phoenix of Madison, Inc., 2001 S. Stoughton Road, Madison, WI 53716)

A common strategy for accommodating reduced dynamic range in the hearing impaired is the use of amplitude compression. In order to address the resulting elevation of the noise floor, an expansion region may be introduced. This study examines the perceptual and acoustic consequences of such nonlinear signal processing of CV and VC monosyllables. Transmitted information for a system of distinctive features as a function of the degree of nonlinearity, S/N ratio, and speaker was evaluated using the SINFA procedure. Acoustic analyses of processed syllables were carried out in an attempt to define articulation index measures that account for the perceptual data.

LL23. Longitudinal changes in syllable discrimination among normal and handicapped school children. Lois L. Elliott and Michael A. Hammer (Audiology and Hearing Impairment, Northwestern University, Evanston, IL 60201)

Cross-sectional age-related differences in discrimination of voice onset time (VOT) [L. L. Elliott, *J. Acoust. Soc. Am.* **80**, 1250–1255 (1986)] and the place-of-articulation (POA) feature [L. L. Elliott *et al.*, *J. Acoust. Soc. Am.* **70**, 669–677 (1981)] when just-noticeable differences among five-foamant synthesized CV syllables are measured using adaptive test procedures were previously described. Results are reported here for children tested *longitudinally* for 3 consecutive years. Discrimination of temporal information (VOT) showed larger changes over time than discrimination of frequency-based information (POA). Also, larger performance differences occurred between normally progressing children and those with learning disabilities for VOT than for POA. Children with learning disabilities generally exhibited poorer auditory discrimination than normals; however, they demonstrated as much longitudinal change in performances as did the normally progressing children (i.e., there were no significant interactions between year of test and normal versus handicapped status). [Supported, in part, by NINCDS and NIH.]

LL24. The role of static spectral properties in vowel identification. James Hillenbrand and Brendan J. McMahon (RIT Research Corporation and Rochester Institute of Technology, 75 Highpower Road, Rochester, NY 14623)

The classic study of Peterson and Barney [*J. Acoust. Soc. Am.* **24**, 175–184 (1952)] demonstrated that vowels in CVC context are identified

with a very high degree of accuracy by human listeners. A number of attempts have been made to classify the Peterson and Barney vowels using some combination of F_0 and the three lowest formant frequencies. Although these attempts have been reasonably successful, the accuracy of the classification algorithms has been well below that of human listeners. Human listeners, however, have access to dynamic (duration and spectral change) as well as static information (F_0 and "target" formant frequencies). The purpose of the present study was to determine the identifiability of the Peterson and Barney vowel set based exclusively on static information. A formant synthesizer was used to generate isolated, steady-state versions of all 1520 vowels (76 speakers \times ten vowels \times two repetitions) in the Peterson and Barney data base using measured values of F_0 and F_1 – F_3 . Preliminary results from a small group of trained listeners suggest that the error rate for the resynthesized stimuli is 3.5–4.5 times greater than the error rate reported in the original Peterson and Barney study. Pilot data also suggest that the identification task may be extremely difficult for some untrained listeners. [Work supported by Rome Air Development Center under contract #F3060285-C-0008.]

LL25. Vowel and consonant judgments are not independent when cued by the same information. D. H. Whalen (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Vowels with ambiguous formant patterns can be perceived as changing from [æ] to [ɛ] as duration decreases. Similarly, ambiguous final consonants can be perceived as [d] when the preceding vowel is long and [t] when it is short. Mermelstein [Percept. Psychophys. 23, 331–336 (1978)] had subjects identify four syllables ("bad," "bat," "bed," and "bet") that were synthesized by varying only vowel duration. He claimed the consonant and vowel judgments were independent, implying distinct vowel and consonant systems. The present replication indicates that Mermelstein's analysis lacked power and that, in fact, the judgments are dependent on each other: Subjects were more likely to report hearing [æ] and [t] together, since more of the vocalic segment duration is attributed to the vowel and/or less is attributed to the consonant. Similarly, subjects are more likely to report hearing [ɛ] and [d] together, since there is less duration attributed to the vowel and/or more attributed to the consonant. This indicates that listeners hear a coherent syllable, in which two distinctions that depend on the same information must compete for that information. [Research supported by NIH Grant HD-01994.]

LL26. Statistical representation of word-initial obstruents: Further data. Karen Forrest, Gary Weismer, and Paul Milenkovic (Speech Motor Control Laboratories, University of Wisconsin—Madison, Madison, WI 53705-2280)

At the 113th Meeting of the Acoustical Society, a procedure was presented for the classification of word-initial obstruents. The procedure involved calculating FFTs at 10-ms intervals from the onset of the obstruent through the third cycle of the following vowel. These FFTs were treated as random probability distributions from which the first four moments (mean, variance, skewness, and kurtosis) were computed; computations were also made on Bark transformed FFTs. Moments calculated over the first 40 ms of voiceless stops discriminated place of articulation with 97% accuracy. Voiceless sibilants were categorized with about 98% accuracy when the Bark transformed moments from the first 20 ms of the fricatives were used. The speech sample used in the previous study only allowed comparison of obstruents across two vowel contexts. In this report, results of the moments analysis of a speech sample are presented in which all voiced and voiceless stops, as well as voiceless sibilants, are paired with six vowels, /i, a, u, æ, ʌ, e/. The efficacy of the moments in discriminating obstruents in this broader context will be discussed. [Work supported by NIH award NS13274.]

LL27. Use of multiple speech dimensions in concept formation by Japanese quail. Keith R. Kluender and Randy L. Diehl (Department of Psychology, University of Texas, Austin, TX 78712)

Japanese quail (*Coturnix coturnix*) have learned to form a category for syllable-initial [d], and can discriminate [d] syllables from those beginning with [b] or [g] across vowel contexts [Kluender *et al.*, Science

(1987)]. Acoustic analysis of the categorized syllables revealed no single feature that could have supported generalization, suggesting that the quail used some combination of stimulus properties to categorize syllables. Current experiments investigate two ways such a category could be formed. First, initial consonant information may be evaluated conditional upon the vowel that follows. Whether quail can learn to categorize syllables in which, for example, positive exemplars are CVs with either a voiced stop and a front vowel, or a voiceless stop and a back vowel is being tested. A second possibility is that phonetic categories are examples of *polymorphous* concepts, with no single necessary or sufficient condition for class membership. Two birds have learned a concept in which three stimulus dimensions of syllables are needed to make a category assignment. The best subject has learned a concept that requires her to discriminate: (1) voiced versus voiceless syllable-initial stops, (2) front versus back vowels, and (3) sex of the talker. Responses to novel stimuli reveal a category structure remarkably similar to that hypothesized for humans. The fact that quail can learn a concept in which multiple cues are clearly required suggests that general mechanisms of audition and perceptual learning are sufficient for acquisition of speech categories. [Work supported by NICHD.]

LL28. Qualitative separateness in children's speech. Susan Nittrouer (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131) and D. H. Whalen (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Acoustic analyses of fricative-vowel (FV) syllables spoken by adults and young children (ages 3 to 7 years) have demonstrated several differences. In children's speech: (1) the gross spectra of /j/ and /s/ segments were more similar than those of adults; (2) fricative F_2 frequencies varied more as a function of vowel context; and (3) the relative F_2 amplitude was greater. These acoustic differences between children's and adults' speech provided an opportunity to test two competing hypotheses of perceptual segmentation. According to one, listeners divide the signal into temporally discrete, context-sensitive allophones that can be used to make inferences about later-arriving segments. The second hypothesis suggests that listeners can separate simultaneously arriving acoustic information into successive phonetic units. In the present set of experiments, brief portions of fricative noise extracted from FV syllables spoken by adults and children to normal adult listeners were presented. Results showed that listeners were less accurate in fricative identification for child speakers, but more accurate in identifying the vowel context from which the fricative noise came. Thus the greater coproduction found in children's speech was more easily recovered by listeners, even though the most prominent phone was more poorly perceived. This finding was taken as support for the second hypothesis. [Work supported by NIH Grants NS-07237 and HD-01994 to Haskins Laboratories.]

LL29. Recognition of speech and recognition of speaker sex: Parallel or concurrent processes? M. O'Kane (School of Information Sciences and Engineering, Canberra College of Advanced Education, Belconnen, A.C.T. 2616, Australia)

If the same phonetic string is spoken by a male speaker and a female speaker, both drawn from a linguistically homogeneous population, then a listener perceives that both speakers are saying the same sounds but generally knows that the speakers are different and, in particular, that the speakers are of a different sex. What is not well understood is whether the listener has to determine the identity of the sex of the speaker in order to determine the identity of the sounds, or whether the two pieces of information are essentially independent and can be determined concurrently, or whether the identity of the sounds can be roughly determined without knowledge of the speaker's sex but fine recognition is possible only after the speaker's sex is known. In this presentation, evidence for and against these various possibilities from a large number of psycholinguistic studies, particularly those dealing with various sex-specific effects in a variety of languages, is surveyed and the implications for the design of automatic speech recognition and understanding systems are discussed.

LL30. Effects of speaker gender on speech recognition by hearing impaired listeners. Angelika Stieren and Winifred Strange (Department of Communicology, University of South Florida, Tampa, FL 33620)

Audiologists and clients often claim that female speech is less intelligible than male speech for subjects with bilateral high-frequency sensorineural hearing losses. To examine the extent to which source characteristics (voice pitch) and gender-related resonance characteristics contribute to these speech recognition difficulties, two men and two women each produced three sublists of the modified rhyme test (MRT). Within gender groups, one speaker had a relatively low voice pitch, one a high pitch, such that the high-pitched male voice was approximately the same as the low-pitched female voice. Stimuli were presented to listeners at 30 dB SL in quiet, in speech noise ($S/N = +12$ dB), and in speech babble ($S/N = +12$ dB). Overall results showed significant differences in identification of consonants across speakers. However, contrary to expectations, performance was worst for the male with the low voice pitch, intermediate for the two female speakers, and best for the high-pitched male. There appeared to be gender-related differences in the type of consonant errors.

LL31. Are female voices more intelligible than male voices? Florian J. Koopmans-van Beinum and Mirjam T. J. Tielen (Institute of Phonetic Sciences, University of Amsterdam, Herengracht 338, 1016 CG Amsterdam, The Netherlands)

Studies on the intelligibility of female compared to male voices, and studies on the relation between acoustic parameters (such as F_0) and speech understanding are rather scarce and indirect. However, particularly in information-providing services, female voices are frequently used, whereas, in speech-technology applications, they are often excluded, probably due to inadequate methods for analyzing female voices. In order to study the relative intelligibility of male and female voices, ten male and ten female speakers were asked to read three phonotactically balanced lists of 50 CVC combinations. Special care was taken with respect to speech level, signal-to-noise ratio, and presentation method using: equivalent peak level, babble noise, and a variation of the Nakatani and Dukes method. For a small number of trained listeners, the intelligibility threshold per speaker is established, providing 20 measuring points per listener. Furthermore, word and phoneme intelligibility per speaker are measured with phonetically balanced sentences and words, at a fixed S/N ratio.

THURSDAY MORNING, 19 NOVEMBER 1987

TUTTLE NORTH ROOM, 9:00 A.M. TO 12:00 NOON

Session MM. Structural Acoustics and Vibration IV: Viscoelastic Polymers: Their Description and Evaluation

Wayne T. Reader, Chairman

Code 1905.2, David W. Taylor Naval Ship R&D Center, Bethesda, Maryland 20084

Chairman's Introduction—9:00

Invited Papers

9:05

MM1. Viscoelasticity of polymers—Consideration of predictive methods for wide frequency response. Raymond E. Wetton (Polymer Laboratories Limited, The Technology Centre, Epinal Way, Loughborough LE11 0QE, United Kingdom)

The nature of viscoelasticity in high polymers is discussed in molecular terms. The factors affecting damping and dynamic modulus are outlined. Many commercial polymers are multiphase and the width and magnitude of molecular relaxation are shown to be related to the compatibility of the components. Master curves showing viscoelastic response over a wide range of frequency can be generated using semiempirical procedures of time-temperature shifting of data. The Williams, Landel, and Ferry (WLF) relation, in particular, is discussed for multi- and single-component polymers. In general, predictions of dynamic response via extrapolation over one or two decades in frequency can be made with reasonable confidence, if enough detailed care is taken. The use of dielectric loss peak loci at high frequencies is proposed to aid these predictions.

9:30

MM2. The temperature dependence of viscoelastic responses above T_g . Robert F. Landel (Jet Propulsion Laboratory, 4800 Oak Grove Drive, Pasadena, CA 91109)

The origins of the WLF equation will be reviewed, along with the role of the temperature dependence of responses in trying to ascertain relaxation mechanisms. Some pitfalls, misuses, and misunderstandings will be described, together with similar comments on so-called T_g 's. The general applicability and utility of the temperature scaling of the time domain will be illustrated with examples from damping, rupture behavior, friction, and crack propagation.

MM3. Molecular basis of the acoustic properties of polyurethanes. Bruce Hartmann, Gilbert F. Lee, and James V. Duffy (Polymer Physics Group, Naval Surface Weapons Center, Silver Spring, MD 20903-5000)

The acoustic properties of polyurethanes as a function of frequency are dominated by a very broad relaxation process in which the polymer makes a transition from the rubbery to the glassy state. It is convenient to use an analytic representation of the data that expresses the complex shear modulus in the form

$$G^* = G_\infty - (G_\infty - G_0) / [1 + (i\omega\tau)^\alpha],$$

where G_0 is the limiting low-frequency modulus, G_∞ is the limiting high-frequency modulus, τ is the relaxation time, and α is a dimensionless parameter, with values between zero and one, that determines the width of the relaxation. It will be shown that τ is related to the glass transition temperature T_g of the soft segment in the polyurethane. In turn, T_g is found to depend on the degree of phase separation that occurs between the chemically incompatible hard and soft segments, as well as the degree of crystallinity developed. These same molecular factors determine G_0 , but G_∞ is found to be essentially constant. Attempts to relate α to the width of the glass transition as measured in a differential scanning calorimeter have been only qualitatively successful. [Work supported by the NSWC Independent Research Program and the Office of Naval Research.]

10:20

MM4. Techniques for noise and signature reduction applications. D. J. Townend (Admiralty Research Establishment, Holton Heath, Poole, Dorset BH16 6JU, United Kingdom)

In recent years, DMA test systems have become commercially available. Prior to this, workers in this field developed equipment to meet their own specific needs. Current commercial equipment is aimed at the thermal analysis market and is seen as complimentary to established techniques, such as DSC and TMA. Great emphasis is placed on temperature control, but, in general, less attention is paid to frequency range and mechanical design. Upper frequency limits are usually around 100 Hz. While low frequencies are ideal for the study of molecular relaxation processes and for routine material characterization, they have little immediate application in the field of noise and signature reduction. Recent work at ARE has resulted in the development of computerized WLF technique; this method is based on a reference frequency as well as a reference temperature. In general (though not essential), the reference point selected is the temperature of peak $\tan \delta$ at a frequency of 1 Hz. Where multiplexed frequency data are available, it is a simple matter to verify that an unknown material will fit the standard equation by comparing measured shifts in peak $\tan \delta$ with computed values based on the standard constants. Signature reduction applications require systems that, in some cases, work at frequencies up to 1 MHz. Computerized predictions based on WLF data, shifted from low-frequency measured data, show very good agreement with high-frequency measured acoustic data, but it would be reassuring to be able to measure dynamic mechanical properties at kilohertz frequencies. Recent improvements to an inertial mass, air-bearing supported system, developed at ARE some 10 years ago, have resulted in a system capable of yielding complex shear modulus data at frequencies up to 10 kHz.

10:45

MM5. Experimental comparison of dynamic mechanical testing facilities at several Navy laboratories. R. N. Capps, R. Y. Ting, and M. E. Quinn (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

Dynamic mechanical spectroscopy provides valuable information on frequency-dependent relaxation mechanisms in high polymers. The Young's modulus and loss tangent are often measured, due to the relative ease of experimental implementation. A variety of techniques is used, which often gives different results for samples that are supposedly of the same materials. A problem exists in interpreting such results in that one can never be sure if such differences are due to measurement techniques or to actual material property variations. Two systems, an automated resonance technique and the Polymer Laboratories DMTA, are now used at several Navy laboratories. To check the internal consistency of results obtained with these systems, a "round-robin" evaluation was performed. Common batches of materials of several different types of rubbers were used to prepare samples for each of three different Navy laboratories. The results of this test will be presented, and factors that can affect the accuracy and reproducibility of dynamic mechanical property measurements will be discussed. Criteria for the applicability of the method of reduced variables will also be treated. [Work supported by NAVSEA.]

11:10

MM6. Standardized presentation of complex modulus. Lynn Rogers (AFWAL/FIBAA Wright-Patterson, Dayton, OH 45433-6553) and Bryce Fowler (CSA Engineering, Inc., 560 San Antonio Road—Suite 101, Palo Alto, CA 94306-4682)

The capabilities of a proposed standard for graphical presentation of complex modulus data for viscoelastic vibration damping materials will be described. A plot of material loss factor versus modulus magnitude indicates data quality; a frequency-temperature nomogram is included on this plot, thereby also making it sufficient for the designer. The more typical reduced frequency plot is also presented, but with experimental temperature and frequency ranges given to assist the designer in avoiding extrapolation of data. The state of the art in modeling temperature shift functions and complex modulus using fractional derivatives will be presented. An interactive computer program for processing a set of complex modulus data will be described. A system to store and retrieve damping material data will be described.

MM7. A computer-based system for representation of damping properties of materials. B. L. Burkewitz (BBN Labs, Inc., Cambridge, MA 02138), J. L. Hamer (E-A-R Division, Cabot Corp., Indianapolis, IN 46268), and C. I. Holmer (Noise Control Technology, Inc., Reston, VA 22091)

The design of a computer-based damping data and design system (DDDS) is described. Important features of the system are: modularity, standardized interfaces between modules based on defined information content, ease of expansion, physically based representation of material dynamic modulus, support for all users from test through analysis, and quality control to treatment design. System architecture and module function are described, followed by a detailed discussion of the material representation module. The input for this module is material test data, and the output is a set of analytic expressions for the frequency and temperature dependence of the material. The software for this module is based on the work of Rogers [J. Rheol. 27, 351-372 (1983)], which provides a rheologically consistent analytic function representation for the dynamic modulus and loss factor. The module is written within *RS/1*, a table-oriented, scientific software system. Damping treatment design software and approaches to system data quality issues are also addressed.

THURSDAY MORNING, 19 NOVEMBER 1987

BRICKELL NORTH ROOM, 9:30 A.M.

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

Technical Advisory Group for ISO/TC 43/SC 1 Noise

W. Melnick, Chairman S12

Ohio State University, University Hospital Clinic, 456 Clinic Drive, Columbus, Ohio 43210

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC 1

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

Standards Committee S12 on Noise. Working group chairs will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC 1 activities will be discussed.

Session NN. Physical Acoustics VII: Acoustic and Seismic Coupling II: Liquid-Solid Interfaces

Michael A. Schoenberg, Chairman
Schlumberger-Doll Research, Old Quarry Road, Ridgefield, Connecticut 06877

Chairman's Introduction—1:00

Invited Papers

1:05

NN1. Seismic anisotropy in the upper oceanic crust—A review. R. A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

Over the past 10 years, there have been four reported observations of seismic azimuthal anisotropy in the upper 2 km of oceanic crust. The most probable cause of anisotropy in the upper crust is the preferred orientation of large-scale fractures that are created during crustal generation at the ocean ridges. Seismic velocities are slow parallel to the spreading direction and fast perpendicular to it. In some instances, clear azimuthally dependent travel-time anomalies are observed and, in others, there is evidence for shear wave splitting. At one site, both shear wave splitting and travel-time anomalies gave consistent results. The challenge in all studies, however, is to separate the effects of anisotropy from the effects of scattering and lateral heterogeneity. Current modeling capability is inadequate to include the effects of both anisotropy and heterogeneity at the scales observed in the seafloor. A definitive study on the effects of anisotropy and heterogeneity that will determine which observations are best for resolving given parameters is needed. In addition, high-quality three-component seismic stations are required to provide the data for particle motion analysis. This may require borehole seismic installations. [Work supported by NSF.]

1:35

NN2. Characterization of anisotropic elastic wave behavior in media with parallel fractures and aligned cracks. Michael Schoenberg (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

Large parallel fractures or aligned microcracks, whether fluid filled, empty, or debris filled, cause specific types of material compliance increases, which are manifest in the anisotropic behavior of elastic waves in such media. The anisotropy attributable to the fracture system may be triclinic (the most general) with six independent compliances, monoclinic with four, orthorhombic with three, or transversely isotropic (rotationally symmetric fractures) with two. The fracture system may be embedded in an anisotropic elastic background with no restrictions on the type of anisotropy. To compute the long wavelength-equivalent moduli of the fractured medium requires, at most, the inversion of two 3×3 matrices. Dilute systems of flat aligned microcracks, assumed on average rotationally symmetric, in an isotropic background yield an equivalent medium of the same form as that of the isotropic medium with large transversely isotropic fractures; i.e., there are two additional parameters due to the presence of the microcracks that play roles in the stress-strain relations of the equivalent medium *identical* to those played by the parameters due to the presence of large fractures. Thus knowledge of the total of four parameters describing the restricted transverse isotropy of such a fractured medium tells nothing of the size or concentration of the aligned fractures, but does contain information as to the overall excess compliance due to the fractures or cracks and their orientation.

2:05

NN3. Borehole modes in a fluid-saturated permeable medium: Numerical and experimental results. Hsui-lin Liu and Kenneth W. Winkler (Schlumberger-Doll Research, Ridgefield, CT 06877)

Stoneley waves propagating in boreholes in permeable materials provide a unique opportunity to study the boundary conditions at a fluid/permeable-solid interface. First, a review of the theory and present frequency domain solutions for Stoneley wave velocity and attenuation is presented. Formation permeability is shown to decrease the Stoneley wave velocity and increase the attenuation compared to the nonpermeable case. This effect is most pronounced at low frequencies. Then the theory is applied to finite size cylinders, such as might occur in laboratory experiments. In this case, two classes of extensional modes, besides the Stoneley mode, are found. One class is the fast wave mode, which is analogous to the ordinary extensional mode. The other class is the slow wave mode, so-called because the zeros are located along the Biot slow wave branch cut. For a highly permeable medium, the slow wave modes can obscure the identification of the Stoneley mode. Also presented is the experimental confirmation of the numerical predictions for Stoneley wave propagation. Compared to conventional ultrasonic techniques, the new Stoneley wave technique allows the performance measurements in a broad frequency band above and below the critical frequency of Biot theory. One application of the results is estimating rock permeability using the borehole Stoneley mode in petroleum prospecting.

NN4. The effects of azimuthal anisotropy on seismic shear wave data. R. M. Alford (Amoco Production Research, P.O. Box 3385, Tulsa, OK 74102)

As a result of seismic field experiments, shear polarization splitting has been identified as a significant impediment to shear wave data quality. The polarization splitting, or shear birefringence, results when a transversely polarized elastic shear wave propagates through an azimuthally anisotropic medium. As a consequence of this observation, the perception of shear wave propagation has been significantly altered and new techniques for acquisition, processing, and interpretation have been developed. In this paper, the anomalous behavior of the data is examined and techniques are developed to compensate for the effects of shear polarization splitting. The corrected data reveal a time-varying delay between two shear polarizations, which is characteristic of shear wave propagation in an azimuthally anisotropic medium. Azimuthal anisotropy may provide another parameter for discriminating rock types and properties, and the possibility of detecting fractures and permeability trends. The realization that shear wave splitting is an important phenomenon helps to explain the erratic and unpredictable nature of shear data quality. New techniques provide an exciting renewal of potential for the exploitation of shear wave seismic data in exploration.

Contributed Papers

3:05

NN5. Coupling of sound with surface waves on fluid-loaded elastic objects described by a generalization of GTD. Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

The geometrical theory of diffraction (GTD) was recently extended to describe surface elastic wave (SEW) contributions of scattering from fluid-loaded spheres and cylinders at high frequencies [P. L. Marston, *J. Acoust. Soc. Am. Suppl. 1* 81, S14 (1987)]. The coupling of an SEW with the acoustic field was described by a complex coefficient G . That analysis gave simple physical approximations: $|G| \approx 8\pi\beta c/c_{sw}$ for spheres and $|G| \approx 8\pi\beta/(\pi ka)^{1/2}$ for cylinders; these were confirmed with other numerical results. The SEW angular attenuation coefficient (due to radiation damping) and phase velocity are β and c_{sw} , while c denotes the sound speed in the fluid, $k = \omega/c$, and a is the object's radius. The present work extends the original analysis of backscattered sound to cases where the SEW only partially circumnavigates the object. If a plane sound wave of amplitude p_i excites an SEW on a cylinder, the sound amplitude radiated by the SEW after traveling an arc length L is $p \approx p_i (a/2r)^{1/2} \times |G/2| \exp(-\beta L/a)$. The observer is a distance r from a virtual source (in the cylinder), from which the radiated rays appear to diverge. A simple approximation of β shows that $\beta \propto$ the ratio of fluid to solid densities. The ratio of the $|G|$ for spheres and cylinders is confirmed by geometrical considerations. [Work supported by ONR.]

3:20

NN6. A Gaussian beam representation of wave fields at a liquid-solid interface. J. J. Wen and M. A. Breazeale (Department of Physics, The University of Tennessee, Knoxville, TN 37996-1200)

A new method of analysis is described to represent the resultant field when a wave beam is subjected to any physical process (e.g., propagation, focusing, reflection, diffraction, etc.). Instead of expressing the field solution as a continuous distribution of plane waves through spatial Fourier analysis, a set of Gaussian beams is employed as the basic unit. The Gauss-

ian beam representation is introduced to replace the traditional plane-wave representation so that the solution is an analytical one rather than a symbolic integral. This replacement is made possible by a computer optimization routine that overcomes problems introduced by use of nonorthogonal functions, and enables the expression of the field solution in a simple closed analytical form—the sum of Gaussian functions. The technique is applied to the description of the reflected field of a bounded beam incident at the Rayleigh angle with very satisfactory results. [Research sponsored by the Science Alliance, a State of Tennessee Center of Excellence.]

3:35

NN7. Saltus-type boundary conditions for the modeling of rough interfaces in acoustic wave problems. Adriannus T. de Hoop, Sijze M. de Vries (Faculty of Electrical Engineering, Delft University of Technology, P.O. Box 5031, 2600 GA Delft, The Netherlands), and Douglas Miller (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

The method of moving volume averages is used to derive boundary conditions of the saltus or jump type for the modeling of acoustic wave interaction across a rough interface between two media. For this purpose, a smooth boundary layer is constructed that contains the rough interface in its interior. The medium in this layer is considered as a mixture of the two media present. On the scale at which the acoustic wave motion is analyzed, the mixture is assumed to have acoustically linear macroscopic properties. The resulting saltus-type boundary conditions express the jumps in the basic quantities that describe the wave motion (pressure or stress and particle velocity) across the boundary layer in terms of the limiting values that these quantities attain at either side of the layer in a linear manner. Conditions of this type are worked out for a rough interface between (a) two fluids, (b) two solids, and (c) a fluid and a solid. In all cases, full anisotropy, including the one that is induced by the interface geometry, is taken into account.

Session OO. Speech Communication VI: Focus Session on the Female Voice

Caroline G. Henton, Chairman

*Linguistics Program, University of California—Davis, Davis, California 95616**Invited Papers*

1:00

OO1. The female voice—Experiments and overview. Gunnar Fant, Christer Gobl, Inger Karlsson, and Qiguang Lin (Department of Speech Communication and Music Acoustics, Royal Institute of Technology (KTH), Box 70014, S-100 44, Stockholm, Sweden)

Results from recent experiments at the KTH directed to the study of aerodynamics and voice source flow will be presented. These include simultaneous recordings of glottal airflow and supraglottal pressure from a Rothenberg mask. A number of females served as subjects producing isolated syllables and, in one instance, complete sentences. For one subject, the subglottal pressure was also recorded. Typical temporal profiles of these measures, including glottal leakage, will be discussed. In a separate study, data were collected on glottal flow waveforms from inverse filtering of the speech pressure wave. A parametrization and resynthesis in terms of the four-parameter LF model has been undertaken, which allows a comparison with a set of male voices. Theoretical studies of source–vocal-tract interaction effects provide a basis for interpretation of results and modeling of female specific characteristics. Finally, a review is presented of accumulated insight in the properties of female speech including factors other than source mechanisms.

1:30

OO2. Glottal airflow and pressure measurements for female and male speakers in soft, normal, and loud voice. E. B. Holmberg, R. E. Hillman (Research Laboratory of Electronics, 36–525, Massachusetts Institute of Technology, Cambridge, MA 02139 and Department of Communication Disorders, Boston University, Boston, MA 02215), and J. S. Perkell (Research Laboratory of Electronics, 36–525, Massachusetts Institute of Technology, Cambridge, MA 02139)

Measurements on the inverse filtered airflow waveform (the “glottal waveform”) and of estimated average transglottal pressure and glottal airflow were made for production of syllable sequences in soft, normal, and loud voice by 20 female and 25 male speakers. Statistical analyses showed that, while there was no significant female–male difference in pressure within any of the loudness conditions, there were significant female–male differences in the glottal waveform parameters that should be relevant to differences in voice source characteristics. In normal and loud voice, female waveforms indicated lower vocal fold closing velocity, lower ac (modulated) flow, and proportionally shorter closed phase of the cycle, suggesting a steeper spectral slope for females. In soft voice, female ac flow was significantly lower than males', but there were no significant differences in female–male closing velocity or relative length of the closed portion, suggesting more similar spectral slopes. The dc (unmodulated) flow did not differ significantly between females and males. Possible implications of the results for female and male voice quality are discussed. [Work supported by NIH.]

2:00

OO3. Physiology of the female larynx. Ingo R. Titze (Voice Acoustics and Biomechanics Laboratory, Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242 and The Recording and Research Center, The Denver Center for the Performing Arts, 1245 Champa Street, Denver, CO 80204)

Anatomically, the female larynx differs from the male in vocal fold length, thickness, angle of the thyroid laminae, resting angle of the glottis, vertical convergence angle in the glottis, and a number of other ways. The vocal fold tissue itself differs with respect to the thickness of the mucosa and fiber composition, e.g., the ratio of connective tissue to muscle tissue. Biomechanically, it has been found that the stress–strain curves are slightly more linear in females than males. Based on these and other known differences in morphology and viscoelastic properties, a scaling procedure is attempted for fundamental frequency, intensity, and other acoustic properties. Some rationale is given also for why females develop pathology at different sites in the larynx, given what is currently known about mechanical stress in the tissues.

2:30-2:40

Break

2:40

OO4. Fact and fiction in the description of female and male pitch. Caroline G. Henton (Linguistics Program, University of California, Davis, Davis, CA 95616)

Female speech has been characterized as high pitched, shrill, overemotional, and swoopy. Pitch is, of course, a relative scale, so it would be equally valid to describe male pitch as low pitched, rumbling, under-expressive, and monotonous. But if "swoopiness" is inherent in female speech, it must be asked whether the effect is one of a comparatively wider range, of greater dynamicity, or of a simple arithmetical misinterpretation. It may be the case that previous accounts have been based on absolute (linear) and, therefore, misleading values in Hertz; whereas the use of the semitone, and thus perceptually more appropriate, scale, indicates that female speech should not be so marked on acoustic bases alone. Reexamination of pitch ranges and dynamicity values reported by others leads to the rejection of some of the elements of the stereotype. New data assembled from five males and five females also indicate that, auditorily, female speech contains similar ratios of pitch range in relation to long-term average pitch, as does male speech. Monotonicity and reduced pitch range, as important ingredients in enhancing stereotypical masculine speech (Terrango, 1974; Bennett and Weinberg, 1974), are reexamined.

3:10

OO5. Acoustic correlates of breathiness: First harmonic amplitude, turbulence noise, and tracheal coupling. Dennis H. Klatt (Room 36-523, Massachusetts Institute of Technology, Cambridge, MA 02139)

A selected sample of reiterant speech has been collected from ten female speakers and six male controls in order to quantify acoustic correlates of perceived breathiness of the female voice, and to contrast these measures with comparable data from males. Two sentences with differing stress patterns were spoken by replacing each syllable by [hV] and by [ʔV], where V = [a,i,ae,o,rr]. Detailed analysis of the [a] data reveals (1) wide variation in the strength of the first harmonic (relative to first formant amplitude), with an average increase of about 6 dB for females relative to males; (2) a greater tendency for the third formant to be excited by noise rather than voicing harmonics in the female population; and (3) indications of tracheal poles and zeros in the spectra of vowels adjacent to voiceless consonants in utterances produced by both genders. These three measures of breathiness tend to be greatest in unstressed syllables and toward the end of an utterance. The acoustic data will be correlated with perceived breathiness of utterances from individual talkers in a preliminary attempt to determine which measure is perceptually most salient; future research using synthesis will systematically investigate perceptual salience. [Work supported, in part, by an NIH grant.]

3:40-4:15

DISCUSSANTS: Sandra F. Disner
R. A. W. Bladon

THURSDAY AFTERNOON, 19 NOVEMBER 1987

UNIVERSITY LECTURE HALL, 1:15 TO 4:00 P.M.

Session PP. Psychological and Physiological Acoustics VII: Masking and Detection

Murray F. Spiegel, Chairman

Bell Communications Research, 435 South Street, Room 2E-252, Morristown, New Jersey 07960

Contributed Papers

1:15

PP1. Profile analysis with time-varying stimuli. David M. Green and Quang T. Nguyen (Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32611)

Detecting a change in spectral shape, when the change occurs at a single spectral locus, requires a simultaneous comparison of the unchanged part of the spectrum with the altered or "signal" part. What are the temporal parameters of the stimulus that influence such comparisons? To consider this question, a complex of 21-equal-amplitude components

was used, ranging from 200-5000 Hz, with equal logarithmic spacing between components. The signal was an increase in level of one component of the multicomponent spectrum. Overall level of the sound varied randomly over a 20-dB range. The signal component and the 20 nonsignal components were amplitude modulated separately. The relative phase of the two modulators was manipulated so that the signal and nonsignal components waxed and waned either in-phase or out-of-phase. For modulation rates faster than about 40 Hz, the thresholds in the two conditions are nearly the same. Thus, for slower rates of modulation, the data suggest that the temporal structure of the spectrum is important, while above 40

Hz, only the power spectrum of the stimulus is critical. [Research supported by the National Institute of Health and the Air Force Office of Scientific Research.]

1:30

PP2. Masker envelopes and simultaneous-masking psychophysical tuning curves. Todd W. Fortune and David A. Nelson (Hearing Research Laboratory, University of Minnesota, 2630 University Avenue S.E., Minneapolis, MN 55414)

Recent reports demonstrate that the temporal envelope of a masking stimulus can influence its effectiveness. To examine the probable effects of masker envelope on psychophysical measures of tuning, simultaneous-masking psychophysical tuning curves (PTCs) were obtained from two normal-hearing listeners. The probe was a 250-ms 1-kHz tone burst presented at 60 dB SPL, which was temporally centered within the 500-ms maskers. Maskers included SAM and QFM tones, narrow-band noise, and pure tones. PTCs were obtained in quiet and in the presence of a continuous broadband noise to mask combination tones. Results indicate that fluctuating envelope maskers are less effective than flat envelope maskers only on the low-frequency side of the PTC. These data suggest that the listener is able to listen for the presence of a probe during periods of low masker energy. Four of the six maskers interacted with the probe producing audible combination tones that could be masked by the broadband noise. Combination tones were most pronounced for the QFM and pure tone maskers, which have similar envelopes. The significance of these data will be discussed in terms of frequency and temporal resolution, particularly with regard to the hearing-impaired population. [Work supported by NINCDS.]

1:45

PP3. Tuning curve widths of infants and adults in simultaneous and nonsimultaneous masking. Nancy Benson Spetner and Lynne Werner Olsho (Department of Otolaryngology, RL-30, University of Washington, Seattle, WA 98195)

Tuning curve widths were estimated using the pulsation threshold (PT) technique for infants, 6-month-olds, and adults. The observer-based psychoacoustic procedure [L. W. Olsho *et al.*, *Dev. Psychol.*, in press (1987)] was the test procedure. Listeners discriminated a train of alternating masker and probe tone bursts from a train of masker tone bursts presented with a continuous probe tone. These stimuli were presented in a background of broadband noise at 20-dB pressure spectrum level. The probe level was 10 dB SL. PTs were obtained at the tuning curve tip and at two points, one above and one below the probe frequency, 10 dB above the tip. The resulting Q_{10} s were compared to those obtained in simultaneous masking for 6-month-olds and adults [L. W. Olsho, *Inf. Behav. Dev.* 8, 371-384 (1985)]. Adult Q_{10} s were significantly greater for PT tuning curves at 500, 1000, and 4000 Hz. Infant PT Q_{10} s were significantly greater than simultaneous masking Q_{10} s at 500 and 1000 Hz, but not at 4000 Hz. [Work supported by NIH.]

2:00

PP4. Phase has no effect on simultaneous or forward masked thresholds. Daniel L. Weber (Department of Psychology, Wright State University, Dayton, OH 45435)

Thresholds for sinusoidal signals at 0.5, 2.0, and 6.0 kHz were measured in a variety of masking conditions using a two-interval, forced-choice procedure. All conditions were performed in the presence of a diotic, white, broadband noise with a spectrum level of 20 dB SPL. All sinusoidal stimuli were presented monaurally in the listener's left ear; all fixed level components were presented at 65 dB SPL. In a determination of the Weber fraction for a 295-ms signal masked by a sinusoid of the same frequency and temporal characteristics, the threshold varied with the phase between masker and signal, as one would predict from the assumption that the task depends only upon the detection of a difference in power (or related dimension) between masker and masker plus signal. In these data, the detection of phase differences, per se, appeared to have no effect on threshold. These data provided a reference for a second condition, in

which the 295-ms signal interval was preceded and followed by 295-ms presentations of a sinusoid at the signal frequency. These "fringe" components had the same phase on all presentations; the signal was added to the masker in quadrature. Signal threshold was measured as a function of the phase angle between the fixed fringe components and the masker. In non-signal intervals, the masker and signal were replaced by a presentation of the fringe component. Presentation of the signal thus produced a change in both phase and level compared to the fringe components. However, thresholds were not improved by the presence of the phase change. Both of the above conditions were repeated for a 20-ms masker and signal, and again showed no evidence that a change in phase improves detection performance. With this shorter signal, it was possible to examine threshold as a function of the phase angle between the signal and a 295-ms forward masker. In forward masking, phase had no effect on signal threshold. [Research supported by NSF.]

2:15

PP5. Masking of short tone bursts of frequency sweeps. David A. Fabry, Margaret F. Cheesman (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455), Neal F. Viemeister, and Dean Schei (Department of Psychology, University of Minnesota, Minneapolis, MN 55455)

Masked thresholds for a short probe tone are greater for sweep-frequency tonal maskers than for stationary sinusoidal maskers at equal SPL, and greatest for sweep rates of 20-30 oct/s [G. F. Smoorenburg and F. Coninx, *Hearing Res.* 3, 301-316 (1980)]. The present study examined possible explanations for this surprising phenomenon and concentrated on the effects of masker level on probe threshold. Masked thresholds were obtained for a 10-ms, 1-kHz tone temporally centered in the 100-ms masker. The maskers were swept in frequency at rates of 0-80 oct/s and passed through 1 kHz at 50 ms. At the lower masker levels, maximum masking occurred at sweep rates of 15-25 oct/s. At higher levels, little influence of sweep rate on masked thresholds was seen. Although there were substantial individual differences in slopes, the growth-of-masking functions had slopes less than one with minimum slopes at 20-25 oct/s. The shallow slopes are a distinguishing feature of sweep-frequency masking and may provide a basis for understanding this important phenomenon. [Supported by NINCDS grant NS12125 and SSHRC-Canada.]

2:30

PP6. Tonal prominence effects on annoyance ratings for business equipment spectra. G. R. Bienvenue (State University of New York, College at New Paltz, New Paltz, NY 12561), M. A. Nobile, and A. C. Balant (IBM Acoustics Laboratory, Poughkeepsie, NY 12602)

While recent researchers have reported Appendix B of ANSI S12.10-1985 to "work well in over 80% of cases" and to be "a vast improvement" over the earlier method, there are two notable difficulties in its implementation. First, when two "audibly prominent" tones occur closely together within a single critical band, they are rated as "not audibly prominent" by the present Appendix, though they produce an audible event. Second, certain sharply falling noise spectra could, by the upward spread of masking, mask a tone that is "audibly prominent." In this case, an inaudible tone would be rated as prominent. For the present study, 15 subjects were presented with tape recorded sounds of varying tone-noise ensembles designed to explore these phenomena. Subjects rated these sounds for their levels of loudness, tonal prominence, and annoyance. The research data are discussed with special attention to their relevance for implementing Appendix B of ANSI S12.10-1985. [Work supported by IBM Corporation.]

2:45

PP7. Modeling subject responses in a reproducible noise masking task. R. H. Gilkey and T. A. Meyer (Signal Detection Laboratory, Central Institute for the Deaf, Saint Louis, MO 63110)

Gilkey and Robinson [*J. Acoust. Soc. Am.* 79, 1499-1510 (1986)] modeled subject responses to individual reproducible noise samples, ran-

domly selected on each trial from a set of 25, in a diotic tone-in-noise detection task. They found that a detector composed of a 50-Hz-wide single-tuned filter centered at the 500-Hz signal frequency, a nonlinearity, and an integrator with a 100- to 200-ms decay constant predicted the data relatively well. Here, a set of 150 noise samples was used, and a more detailed analysis of the parameter space of the detector was performed. The results of this study are similar to their results, although the bandwidths of the filters are smaller (26–49 Hz) and the decay constants are shorter (39–100 ms). "Spectral weighting functions" were derived from linear combinations of the outputs of seven detectors, each at a different center frequency. These functions are more consistent than those of Gilkey and Robinson and also indicate that subjects compare information across frequency regions. These functions can be described as the difference between two Gaussian-shaped weighting functions. [Supported by NSF and AFOSR.]

3:00

PP8. Detection of infratone repetition of frozen noise: Singularity recognition or pattern recognition? Brad S. Brubaker and Richard M. Warren (Department of Psychology, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

Listeners can readily detect iteration of "frozen" Gaussian noise segments with durations up to 1 s [N. Guttman and B. Julesz, *J. Acoust. Soc. Am.* **35**, 610 (1963)]. Is detection of repetition of these complex waveforms based upon recognition of the reoccurrence of unique features or singularities, or upon a more holistic recognition of the pattern formed by these events? To answer this question, frozen noise segments were divided into three sections of equal duration (A, B, C) that were reassembled and then repeated to form two periodic sounds $(ABC)_n$ and $(ACB)_n$. This manipulation changed the temporal arrangement between sections but preserved singularities and repetition rate. Untrained listeners heard a series of sequence bursts consisting of either one arrangement $[(ABC)_n, (ABC)_n, (ABC)_n, \dots]$ or two alternating arrangements $[(ACB)_n, (ABC)_n, (ACB)_n, (ABC)_n, \dots]$ and judged whether successive bursts were the same or different. Discrimination was possible when the duration of the entire iterated pattern $(A + B + C)$ was 900 ms or less, indicating that a holistic recognition of patterns operates up to the limit of echoic storage. [Work supported by AFOSR.]

3:15

PP9. Detection and recognition of complex stimuli. David A. Ansley and Daniel L. Weber (Department of Psychology, Wright State University, Dayton, OH 45435)

Green *et al.* [*J. Acoust. Soc. Am.* **62**, 948–954 (1977)] have demonstrated that a subject's ability to detect one of m possible signals in a yes/no task can be used to predict the subject's ability to identify which of the m possible signals was present when the signals are four sinusoids well separated in frequency. This study examines detection and recognition when the signals are four complex stimuli presented against a background of white noise (spectrum level 20 dB SPL). Each complex stimulus consists of a temporal sequence of seven sinusoids. Each sinusoid differs in frequency from the others within a pattern, and each is presented for 100 ms at the same sensation level. The patterns differ in the sequence of the frequencies of the component sinusoids. For example, the frequencies in

the second pattern are monotonically decreasing, whereas those in the third pattern are monotonically increasing. Six experimental conditions are examined, obtained by varying the frequency separation between components and patterns. The condition with the greatest frequency separation is one in which there is a one-fourth octave gap between patterns; the condition with the least frequency separation is one in which all four patterns are composed from a single set of frequencies. Except for the last condition, successive patterns within each condition are centered around 700, 1000, 1400, and 2000 Hz, respectively. Subjects' performance in the recognition task is at the level predicted on the basis of their detection performance, in those conditions with greater frequency separation between patterns. Recognition performance falls below the predicted level in those conditions with little or no frequency separation. [Research supported by AFOSR.]

3:30

PP10. Forced-choice procedures can yield contaminated thresholds: Masker response bias. Murray F. Spiegel (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

Early measurements of human performance were plagued by response biases, such as a preference for choosing "yes" in a yes/no signal-detection task, or a tendency to choose the second of two sounds as louder, longer, or higher. Signal detection theory, as well as forced-choice and adaptive procedures, have largely eliminated the influence of response biases while also streamlining the data collection process. These procedures are adequate when the set of maskers are *indistinguishable*. However, adaptive procedures, when combined with forced-choice methods, are sensitive to response bias when the various maskers are distinguishable. The response bias is a preference for particular *maskers*, when maskers are randomized within a trial. Thresholds can be seriously contaminated; depending on the magnitude of the bias, subjects can appear to be responding perfectly to inaudible signals. Procedures that avoid the problem resulting from this form of response bias are not as efficient as adaptive, forced-choice procedures, and often entail more traditional methods of testing.

3:45

PP11. A context-free rubber band model of scaling auditory dimensions. Irwin Pollack (Mental Health Research Institute, University of Michigan, Ann Arbor, MI 48109)

Auditory signals varying in frequency, duration, and/or amplitude were presented to listeners. The listener's task was to mark a linear scale to rate the relative position of one sound on a scale demarcated by the other two. The signals that defined the ends of the scale were varied from trial-to-trial. The ratings are well-predicted by assuming that the ratio of scale distances is invariant with the particular sounds that define the scale. Stated otherwise, the rating scale is conceived in terms of positions upon a rubber band. Different scales, defined by different end stimuli, result in local compression and extension of the band, but relative distances among intermediate ratings are unchanged by scale changes. The model's apparently excellent performance may simply result from the powerful constraint of monotonicity in the scaling of unidimensional signals.

THURSDAY PM

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

D. Johnson, Chairman S1

Larson-Davis Laboratories, 280 South Main, Pleasant Grove, Utah 84062

Standards Committee S1 on Acoustics. Working group chairs 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 Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, for which S1 and S3 serve as the U.S. Technical Advisory Groups (TAGs), will be discussed. The Chairs of the TAGs for ISO/TC 43 (H. E. von Gierke) and IEC/TC 29 (V. Nedzelnitsky) will report on current activities of these Technical Committees.

Session QQ. Education in Acoustics II: Education via Satellite and Video Tapes

Lawrence A. Crum, Chairman

National Center for Physical Acoustics, P.O. Box 847, University, Mississippi 38677

Invited Papers

1:30

QQ1. A satellite classroom for advanced acoustic studies. Richard Stern (Applied Research Laboratory, Penn State University, P.O. Box 30, State College, PA 16804)

The Graduate Program in Acoustics, Penn State University, is offering a Master of Engineering Degree in Acoustics via satellite to the Navy Undersea Weapons Engineering Station (NUWES), Keyport, Washington. The program will take 2 years to complete and consists of both televised and on-site instruction. Progress of the program and lessons learned will be discussed.

1:55

QQ2. Graduate continuing education by satellite. Lionel V. Baldwin (National Technological University, P.O. Box 700, Fort Collins, CO 80522)

Progress in telecommunications technologies presents the engineering profession with exciting new ways to integrate advanced study with the job. Ten years ago the U.S. colleges of engineering that operate regional ITV systems started planning for a common goal: to increase the national effectiveness of continuing education of engineers. Two new institutions, the National Technological University (NTU) and the Association for Media-Based Continuing Education (AMCEE) that coordinate credit and continuing education programs, respectively, are now accelerating the introduction of linkages between engineering faculty and

engineers at work via a satellite. Both NTU and AMCEE began regular satellite delivery of advanced technical education in August 1985. As we enter the third year of operations, NTU will serve more than 3200 enrollees for academic credit instruction during 1987-1988; AMCEE will provide over a thousand hours of noncredit, live-and-interactive programming. The service operates in GSTAR-I with a modern Kuband transponder, which broadcasts two channels of full-motion, color video throughout the day and evening of each workday. Eighteen universities currently operate uplinks to provide live programming over the networks and, in addition, three corporate studios have uplinks and regularly originate instruction.

2:20

QQ3. Using video tapes and audio tapes on and off the campus. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The use of prerecorded presentations offers several advantages to the teacher. One of the most attractive is to present demonstrations of acoustical phenomena that are difficult to implement in real time, especially in a remote classroom. Another is to substitute animated text for still text (the so-called "animated blackboard"). Excerpts are presented from several audio and video tapes and how they can be used to enhance acoustics instruction both on and off campus is discussed.

THURSDAY AFTERNOON, 19 NOVEMBER 1987 BOUGAINVILLEA ROOM, 1:30 TO 4:05 P.M.

Session RR. Engineering Acoustics IV: Transducers, Waveguides, and Structures

Theodore A. Henriquez, Chairman

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

Chairman's Introduction—1:30

Contributed Papers

1:35

RR1. Distortionless piezoelectric transducers with nonuniform electroelastic parameters. Dov Hazony and Justo R. Raya (Department of Electrical Engineering and Applied Physics, Case Western Reserve University, Cleveland, OH 44106)

Of concern are the generation and detection of sound by piezoelectric transducers with nonuniform material properties along the principal axis. It is shown that subject to conditions of distortionless propagation [V. Burke, R. J. Duffin, and D. Hazony, "Distortionless Wave Propagation in Inhomogeneous Media and Transmission Lines," *Q. Appl. Math.* 183-194 (July 1976)] it is possible to associate a circuit model with the transducer similar to the Mason model [W. P. Mason, *Electromechanical Transducers and Wave Filters* (Van Nostrand, New York, 1948)]. This work follows the development of spherical shell piezoelectric transducers [D. Hazony, "Nonuniformly Poled Spherical Shell Piezoelectric Transducers," *J. Acoust. Soc. Am.* 81, 1624-1627 (1987)]. [Work supported in part by Tecsonics, Inc.]

1:50

RR2. Optimizing the performance of piezoelectric drivers using stepped horns. A. Bangviwat and R. D. Finch (Department of Mechanical Engineering, University of Houston, Houston, TX 77004)

An analysis is presented for the design of piezoelectric transducers, in a sandwich configuration using a stepped horn as particle velocity amplifier. The condition for maximum power delivery to the load impedance is established in terms of the parameters of the horn and the piezoelectric material. Analytical solutions can be found readily if one of the horn lengths is chosen. If this is done, then there are two area ratios that satisfy the maximum power condition. The method of false position that is suitable for finding solutions numerically in a general case by using a computer is also described.

2:05

RR3. Interpreting piezoelectric oscillator dissipation effects in terms of the underlying materials viscoelastic relaxation phenomena. R. Lowell Smith (Texas Research Institute, 9063 Bee Caves Road, Austin, TX 78733)

Equivalent circuit modeling has long been recognized as a valuable tool for describing transduction processes and the behavior of transducers. The formalism has been generalized to represent distributed properties, mode coupling, and the implications of structural complexity, mechanical loading, and electrical tuning. Dissipation or loss phenomena can also be accommodated by equivalent circuit methods. It is generally recognized that interpreting all relevant elastic, dielectric, and piezoelectric coefficients as complex numbers is an appropriate way to represent losses. All too often, the scope of this approach discourages the development of detailed results. This paper analyzes the dissipation modeling problem in terms of relaxation processes, a phenomenological approach that gives algebraic form to certain groupings of the electromechanical coefficients. The implication is that sound speed, coupling coefficients, materials moduli, etc., are frequency dependent. A mass-loaded longitudinal vibrator with a segmented piezoelectric cylinder is analyzed as an example of this model-parameter dispersion problem. Of particular interest is the effect of the adhesive joints on the oscillator admittance spectrum over a wide range of frequencies.

2:20

RR4. Development of spherical and cylindrical polarization for three-dimensional piezoelectric finite elements. P. Tierce,^{a)} K. Anifrani,^{b)} J. C. Debus, J. C. Decarpigny (Institut Supérieur d'Electronique du Nord, 41 Bd Vauban, 59046 Lille Cedex, France), and D. Boucher (Groupe d'Etude et de Recherche en Detection Sous Marine, Le Brusac, 83140 Six Fours Les Plages, France)

Three-dimensional analysis of spherical or cylindrical transducers with axially or radially polarized piezoelectric ceramics is rather difficult because it needs a large number of classical elements to take into account the correct polarization. A new type of piezoelectric finite element has been developed in the ATILA code [J. N. Decarpigny *et al.*, J. Acoust. Soc. Am. **78**, 1499 (1985)] to obtain a good representation of the polarization. This paper describes these elements and analyzes the results (in air resonance modes, electrical characteristics, and mode shapes) obtained on some examples.^{*)} Currently at SINAPTEC, 41 Bd Vauban, 59000 Lille, France.

2:35

RR5. Use of special waveforms for optimum efficiency of high-intensity sound sources. Frederic G. Pla (Sverdrup Technology, Inc., NASA LeRC Group, 16530 Commerce Court, P. O. Box 30650, Midpark Branch, Middleburg Heights, OH 44130) and Gerhard Reethof (Noise Control Laboratory, 157 Hammond Building, University Park, PA 16802)

High-intensity sound sources, such as sirens, have received much attention in the recent past due to renewed interest in industrial applications of high-intensity acoustics. As a result of the very high sound pressure levels required (155–165 dB), finite amplitude effects must be taken into account in the design of sound generators. A time domain solution of the second-order nonlinear wave equation is used to predict the behavior of initially nonsinusoidal plane waves, and is compared with a frequency-domain approach. Results for initially sinusoidal, rectangular, and inverse-shock waves are presented. It is shown that the shock formation distance for an initially inverse shock wave is π times the shock formation distance for an initially sinusoidal wave, and that the wave distortion actually results in an amplification of the fundamental, thus increasing the efficiency of the sound generation process. The consequences of wave distortion on several practical high-intensity sound sources is discussed.

2:50

RR6. The distortion spectrum of the fractionally addressed digital oscillator. W. M. Hartmann (Physics Department, Michigan State University, E. Lansing, MI 48824)

The technique of fractional addressing permits one to construct a digital oscillator with arbitrarily high frequency resolution. However, the technique introduces distortion. The distortion power spectrum may be calculated exactly by borrowing mathematical methods from crystallography, such that the crystallographic unit cell length equals the denominator of the irreducible fractional part of the address increment. The calculation exhibits remarkable mathematical symmetries, which lead to simple closed-form expressions for the levels of the components in the distortion spectrum and for the total rms distortion. The expressions show how the distortion may be minimized or possibly employed as an alternative to digital FM in the synthesis of complex tones. [Work supported by the NIH.]

3:05

RR7. Analytical modeling of an active noise control system in a duct. M. L. Munjal and L. J. Eriksson (Corporate Research Department, Nelson Industries, Inc., P. O. Box 600, Stoughton, WI 53589)

Making use of the transfer matrices and electroacoustic analogies, a standing wave analysis of the active noise control system in a duct is presented incorporating the characteristics of the primary source, as well as the auxiliary source. Analytical expressions have thus been derived for the ratio of the two source pressures and certain other ratios or relations of interest for complete cancellation of noise downstream of the auxiliary source. These expressions are then compared with those obtained from transfer functions of the various blocks constituting the block diagram of the entire system. In the process, equivalence has been established between the different entities involved in the two approaches, one of which is used in acoustics and the other in system theory. In particular, it has been shown that the pressure generated by a source against a load impedance can be looked upon as a sum of two pressure waves, one gener-

ated by the source against the characteristic impedance and the other by reflecting the rearward wave (incident on the source) off the source impedance. This principle is seminal in linking the two approaches.

3:20

RR8. On the nature of acoustic singularities arising from the coincidence of flow and sound sources. L. M. B. C. Campos (Instituto Superior Técnico, 1096 Lisbon Codex, Portugal)

The coincidence that flow and acoustic sources can occur in important acoustic problems, e.g., a spherical wave in a conical duct containing an incompressible mean flow, is an example of flow and sound sources collocated at the vertex of the cone; the cone is the particular case $n = 1$ of the power-law ducts of cross-sectional area $S(x) \sim x^{2n}$, which exhibit, for all real n , coincidence of flow and sound sources or sinks at $x = 0$ or $x = \infty$. It is shown that the coincidence of flow and sound sources can lead, for the power law ducts, to algebraic singularities for the phase, and essential singularities for the acoustic field; a transformation involving the solution of a particular Riccati equation is used to account for the essential singularity, and regularize the acoustic problem. Exact solutions are obtained for sound in parabolic and hyperbolic nozzles, and these are used to study the acoustic field in the ray, asymptotic and compactness approximations, compare acoustic velocity and pressure, discuss equipartition and biasing of kinetic and compression energies, and conservation and nonconservation of wave action. The duality principle for horns [R. W. Pyle, J. Acoust. Soc. Am. **37**, 1187 A (1965)] is shown to fail, and have no simple extension, for nozzles.

3:35

RR9. Whispering gallery resonances on solid elastic spheroids. A. Nagl (Department of Physics, Catholic University, Washington, DC 20064), M. F. Werby (Naval Ocean R&D Activity, Code 221, NSTL Station, MS 39529), H. Überall (David W. Taylor Naval Ship R&D Center, Annapolis, MD 21402 and Department of Physics, Catholic University, Washington, DC 20064), and J. W. Dickey (David W. Taylor Naval Ship R&D Center, Annapolis, MD 21402)

Using the NORDA T matrix code, backscattering amplitudes versus frequency have been calculated for the axial incidence of a plane acoustic wave in water on solid tungsten carbide spheroids with a large variety of aspect ratios. Besides the usual series of broad Rayleigh wave resonances, series of narrow whispering gallery resonances are observed and investigated. The dominant mode number of each of these resonances is identified by an interpretation of the bistatic scattering pattern. For increasing aspect ratios, it is noted that an increasing number of the lowest-order whispering gallery resonances fail to be excited. A possible explanation of this phenomenon is discussed, based on phase matching arguments for the circumferential propagation of surface waves.

3:50

RR10. Sound radiation from beams under the action of moving line forces. R. F. Keltie and H. Peng (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910)

The topic of sound radiation from beams under the action of harmonic line forces moving at subsonic speeds ($M < 1$) is studied. The nondimensional sound power expressions are obtained through the integration of the surface intensity distribution over the entire beam. An asymptotic expression for the sound power in the low-frequency region is derived depending upon the characteristics of the fluid loading and the spatial extent of the applied forces. Numerical integrations have been performed to determine the effects on the radiated sound power of the Mach number M , the acoustic length of line force $K_0 L$, and the wavenumber ratio γ . The results show that: For beams under heavy fluid loading, the effect of the speed of the moving force is not pronounced; while for beams under light fluid loading, the unique coincidence radiation peak at $\gamma \sim 1$ for a stationary force ($M = 0$) is split into two coincidence peaks (located in the frequency region $\gamma < 1$ and $\gamma > 1$, respectively) due to the effects of the Doppler shift. The values of $K_0 L$ that suppress the coincidence peaks are also changed due to the motion of the line force.

Session SS. Noise V and Bioresponse to Vibration II: Effects of Man-Made Noise on Animals and Animal Communication

William C. Cummings, Chairman

Oceanographic Consultants, 5948 Eton Court, San Diego, California 92122

Chairman's Introduction—1:30

Invited Papers

1:35

SS1. Review of studies on the effects of man-induced noise on marine mammals of the Bering, Chukchi, and Beaufort Seas and how the results have been applied to federal offshore oil and gas management decisions. Jerry L. Imm and Cleveland J. Cowles (Minerals Management Service, Alaska OCS Region, Anchorage, AK 99508-4302)

The U.S. Minerals Management Service has been funding and managing studies of the potential effects of man-induced noise on marine mammals, particularly endangered cetaceans, in the Bering, Chukchi, and Beaufort Seas since 1978, and has expended approximately \$5,000,000 toward those efforts. The purpose of such studies is to provide information needed for informed decision making pertaining to environmentally sound leasing and management of offshore oil and gas development on the Alaska OCS. Many of the noise/marine mammal interaction studies have been used to establish or modify lease terms or regulations for offshore operations in federal lease areas. Also, these results have been important in resolution of litigation pertaining to OCS oil and gas leasing/exploration. Specific case examples of how results have been applied will be presented and projection of future Alaska information needs in this discipline will be discussed.

2:00

SS2. Potential impact on sea turtles, dolphins, and fishes of explosives used in offshore platform removals. Edward F. Klima, Gregg R. Gitschlag, Maurice L. Renaud (National Marine Fisheries Service, 4700 Avenue U, Galveston, TX 77551-5997), and William C. Cummings (Oceanographic Consultants, 5948 Eton Court, San Diego, CA 92122)

A high incidence of strandings of sea turtles, bottlenose dolphins, and fishes was recorded on the beaches in the northwestern Gulf of Mexico during March–June 1986, when explosives were used to remove several oil platforms in adjacent offshore waters. Prior to, during, and after March–June of the following year, 1987, strandings for the same area were significantly lower, or negligible. Recovery locations of drift bottles released at the site of the explosions were correlated with some of the strandings. Wild turtles and bottlenose dolphins were observed in the vicinity of the platforms during removal operations. Abnormal (stunted) Kemp's ridley and loggerhead sea turtles were placed at various distances from platforms and exposed to explosions. Preliminary results from exposure studies, including the death of a Kemp's ridley 50 yards from the explosion and an unconscious loggerhead turtle approximately 1000 yards from the explosion, are described. Allowing for in-bottom loss, received sound-pressure levels were calculated to be 235 dB (50 yards) and greater than 200 dB where a caged loggerhead was found unconscious.

2:25

SS3. A review of noise effects studies from the U.S. west coast. Dilworth W. Chamberlain (ARCO, Environmental Protection, 515 South Flower Street, Los Angeles, CA 90071)

Concern over the possibility of effects from geophysical activity on marine organisms has recently generated several research studies in California. Results of effects studies by compressed air releases from airguns (seismic energy releases) on fish dispersion and upon physical effects to fish have been obtained. This information includes the physical effects of a simulated, towed geoseismic array on northern anchovy (*Engraulis mordax*) eggs, larvae, and adults. Other studies have produced data about the schooling behavior of rockfish (*Sebastes spp.*) exposed to airgun releases. Additional research addressing the effects of seismic energy releases on Dungeness crab (*Cancer magister*) eggs and larvae and on rockfish behavior are being planned for the waters of California and Washington.

2:50

SS4. The influence of sound propagation conditions on the behavioral response of whales to underwater industrial noise. Charles I. Malme and Paul R. Miles (Bolt Beranek and Newman Laboratories, Inc., 10 Moulton Street, Cambridge, MA 02238)

Results of recent studies involving gray, humpback, and bowhead whales show that whales tend to avoid areas with high underwater noise levels. Data obtained from whale behavioral observations during controlled

exposure to representative industrial noise sources permit determination of the probability of avoidance P_a of the source region as a function of the noise level L_n . The zone of influence of a source may be defined as the region where $P_a > 0.5$. While the L_n required to produce this degree of avoidance has been found to depend on whale species and source type, some generalizations may be made. For low-frequency continuous noise, 50% of the whales exposed have been observed to avoid regions where the overall L_n is higher than 115 to 125 dB (*re*: 1 μ Pa). Sound transmission conditions at a specific site determine the distance from the source, where L_n falls below the $P_a = 0.5$ criterion level. The zone of influence thus has been found to vary considerably for the test sites investigated. For example, a drillship operating at a test site off the Alaskan Beaufort coast has an estimated zone of influence radius of 4 km, but, off the coast of California, the estimated zone of influence radius is reduced to 1 km. [Work sponsored by the Minerals Management Service.]

3:15

SS5. Reactions of bowhead whales to drilling and dredging noise in the Canadian Beaufort Sea. W. John Richardson (LGL Ltd., Environmental Research Associates, P.O. Box 280, King City, Ontario L0G 1K0, Canada), Bernd Würsig (Moss Landing Marine Laboratories, P.O. Box 223, Moss Landing, CA 95039), and Charles R. Greene, Jr. (Greeneridge Sciences, Inc., 5276 Hollister Avenue, Suite 408, Santa Barbara, CA 93111)

Behavioral reactions of bowhead whales, *Balaena mysticetus*, to seven 30- to 40-min underwater playbacks of recorded drillship and dredge noise were determined in 1982-1984. Water depths were 10-150 m with little or no ice. Some (but not all) bowheads oriented away when received noise levels and spectra resembled those several kilometers from actual drillships and dredges. During some playback tests, call rates decreased, feeding ceased, and cycles of surfacing, respiration, and diving changed. Sensitivity of various whales to noise differed. Roughly half responded to received noise levels of about 115 dB *re*: 1 μ Pa broadband, or about 110 dB in one $\frac{1}{2}$ -oct band (20-30 dB above ambient). Such levels occurred 3-11 km from a drillship and dredge in the Canadian Beaufort Sea. Bowheads occasionally were seen less than 5 km from actual drillships and dredges, where received noise levels were at least as high as during brief playbacks. Thus some bowheads may habituate to prolonged noise exposure. Alternatively, only the least-sensitive individuals may occur less than 5 km from drillships and dredges. [Work supported by U.S. Minerals Management Service.]

3:40-3:45

Break

Contributed Papers

3:45

SS6. Effects of noise of a proposed ocean thermal energy conversion plant on marine animals—A preliminary estimate. Robert S. Gales and Sue E. Moore (SEACO, Inc., 2845-D Nimitz Boulevard, San Diego, CA 92106), William A. Friedl (Naval Ocean Systems Center, Kailua, Hawaii 96734), and James Rucker (National Oceanographic and Atmospheric Administration, Washington, DC 20235)

An analysis is presented of the predicted underwater noise from a proposed 40-MW ocean thermal energy conversion (OTEC) plant to be located approximately $\frac{1}{2}$ mile offshore of Kahe Point, Oahu, Hawaii, and its potential effects on the marine animals of the area. The required total warm and cold seawater flow of approximately 1.6 million liters/min involves 16 pumps driven by a total of 17 000 horsepower. Earlier studies [C P. Janota and D. E. Thompson, *J. Acoust. Soc. Am.* **74**, 256-266 (1983)] calculated the noise generated by various proposed OTEC systems, and compared it with actual measurements on a 1-MW research system (OTEC-1) operating off Hawaii. These analytic procedures are utilized to predict the noise for the 40-MW plant. Underwater noise generated during the construction phase is expected to exceed the plant noise, but for a relatively short period. General auditory perception and expected responses of cetaceans and fishes of the area to both plant operation and construction noises are discussed. The most likely behavioral responses include temporary displacement from the area of the plant and partial masking of communicative signals. [Work supported by NOAA.]

4:00

SS7. Drill-site sounds and bowhead whale calls. Charles R. Greene (Greeneridge Sciences, Inc., 5276 Hollister Avenue, Suite 408, Santa Barbara, CA 93111)

Bowhead whales, an endangered species, migrate westward along Alaska's north coast during September to October, when ice conditions are best for oil explorations on the outer continental shelf. Concern exists that marine activity sounds may adversely influence whale behavior. Underwater sounds were measured in the fall of 1986 near two drillship operations 20 and 35 km offshore, northwest of Kaktovik, Alaska, with water depths of 34-35 m. Sound levels at range 0.2 km, 20-1000 Hz, for three support ships underway at 10-12 kn (standard speed) were 130, 137, and 145 dB *re*: 1 μ Pa. The level from the strongest ice-breaking source, same frequency band and range, was 148 dB. Corresponding levels from the drillship were 135-136 dB during drilling and cleaning and 130 dB during tripping. For measurements within 0.4 km, sound levels were highest in $\frac{1}{2}$ -oct bands from 50-100 Hz. Hourly measurements for 20 days at 11 km from one site revealed a median level of 114 dB in the 20-1000-Hz band, with 5th and 95th percentile levels of 108 and 125 dB, respectively. Dominant sounds at that distance came from the drillship and support ships. Continuous monitoring during 20 days recorded 206 bowhead calls, with 29 of them located 9.5-24 km from the drillship. [Work supported by Shell Western E&P, Inc.]

4:15

SS8. Using sounds to control the movements of sea otters. R. W. Davis, F. W. Awbrey, and T. M. Williams (Sea World Research Institute, Hubbs Marine Research Center, 1700 South Shores Road, San Diego, CA 92109)

Field tests were conducted in Prince William Sound, Alaska, to determine whether the playback of artificial sounds, killer whale calls, or sea otter pup calls could be used to control the movements of sea otters (*Enhydra lutris*). Tests were conducted on three groups of five adult otters that were captured and placed in an enclosed lagoon (180 m long x 40 m wide). Sounds were broadcast in air and underwater from a platform in the center of the lagoon. The behavior and distance of the otters from the platform were determined from video recordings obtained from a remote-controlled video camera overlooking the lagoon. The results showed that sea otters were not repelled by loud (SPL = 120 dB at 1 m), obnoxious sounds in air such as a warble tone (frequency-modulated sinusoid centered at 1 kHz) or air horns and narrow-band pulses at frequencies of 500 Hz-20 kHz projected underwater. However, the otters were attracted to the aerial playback of sea otter pup calls and repelled by the aerial and underwater playback of killer whale calls. Psychoacoustic stimuli appear to have the best chance of influencing the movements of wild sea otters. [This research was supported by Minerals Management Service, Contract No. 14-12-0001-30256.]

4:30

SS9. Effects of aircraft noise on Pacific black brant and other geese in Alaska. David H. Ward, Dirk V. Derksen (U.S. Fish and Wildlife Service, 1011 East Tudor Road, Anchorage, AK 99503), and Paul D. Schomer (U.S. Army Construction Engineering Research Laboratory, P. O. Box 4005, Champaign, IL 61820)

In 1987, a study of the effects of aircraft noise on the behavior, distribution, and habitat use of Pacific black brant, Canadian geese, and em-

peror geese was initiated on the Alaska Peninsula and on the North Slope of Alaska. The objectives were to: (1) describe the behavioral responses of geese as an effect of noise from aircraft overflights; (2) record and examine noise associated with experimental overflights of fixed- and rotary-wing aircraft and specific altitudes, airspeeds, and environmental conditions; (3) secure baseline noise levels for routine flights of nonexperimental aircraft; and (4) provide recommendations to government agencies for reduction or mitigation of any adverse impacts associated with aircraft noise to these populations of geese. Behavior of geese was monitored from remote locations prior to, during, and following level overflights along prescribed routes. Concurrently, 1/3-oct sound exposure and maximum A-weighted and sound levels of each aircraft were measured with a real time analyzer. Here, the design and preliminary results of the first of a multiyear study will be discussed.

4:45

SS10. Acoustic communication in *P. Tigris*. Harry Hollien (IASCP, University of Florida, Gainesville, FL 32611)

While no other species exhibits the extensive language systems of humans, a number are seen to pass information by one or more behaviors. Prominent among the observed signals are those of an acoustic nature; indeed, some subspecies of birds demonstrate fairly complex communicative networks. Among the *Panthera*, *P. Tigris* is perhaps the most vocal. Extensive S/R experimentation with seven captive tigers and subjective observation of nine others (plus evaluation of a few calls recorded in the field), have resulted in classification of ten meaningful utterances plus two others that may be information bearing (not yet "decoded"). This system appears to be somewhat more complex than that exhibited by *Tursiops T.* (and certainly more so than that of the common dolphin). The second stage of this project involves playback of recorded and synthesized calls in order to confirm (or not) the behaviors noted.

THURSDAY AFTERNOON, 19 NOVEMBER 1987

TUTTLE NORTH ROOM, 1:30 TO 5:00 P.M.

Session TT. Structural Acoustics and Vibration V: Viscoelastic Materials: Their Characterization and Application

Fred Schloss, Chairman
Wilcoxon Research, 2096 Gaither Road, Rockville, Maryland 20850

Chairman's Introduction—1:30

1:35-2:00

TT1. Panel Discussion: Characterization of Viscoelastic Polymers

Wayne T. Reader, Moderator

- PANEL MEMBERS: B. L. Burkewitz
 R. J. Capps
 B. Hartmann
 R. F. Landel
 L. Rogers
 D. J. Townend
 R. J. Wetton

2:00

TT2. Structure and acoustical properties of PTMEG polyurethanes. Corley M. Thompson (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337) and Karen L. Gebert (PCR, Inc., P. O. Box 1466, Gainesville, FL 328602)

Polyurethanes are widely used in underwater acoustical devices because of their convenience and attractive physical properties. Rarely, however, have these polyurethanes been designed for the specific application. More typically, a commercial material is used in both sound-transmission and sound-absorption applications, usually with little understanding of structural parameters that control the acoustical properties. This paper will show that the acoustical properties of these polyurethanes may be modified by changing the size of the PTMEG softblock, and that adjustment of the softblock components is ineffective in producing such changes. Evidence will be presented that the structural mechanism for the correlation between acoustical properties and softblock size is the decreasing solubility of the hardblock material with increasing softblock molecular weight. This decreased solubility results in a more pure softblock and in lower dynamic losses and lower specific acoustic impedance. The evidence for this is the regular change in the softblock's glass transition temperature with molecular weight. The acoustical ramifications of these relationships will be discussed.

2:15

TT3. Relationship between polyurethane composition and viscoelastic properties of model urethane systems. R. N. Capps, G. M. Stack, E. M. Dodd (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337), and E. Y. Chang (American Cyanamid, Bridgewater, NJ 08807)

A study was performed to determine the relationship between polyurethane composition and viscoelastic properties in a series of model urethane compounds. The compositional variables studied included the nature of the soft segment, amount of hard segment, and cure stoichiometry. Model urethanes were prepared using the aliphatic diisocyanate, metatetramethylene diisocyanate (m-TMXDI). This aliphatic isocyanate is less reactive than commonly used aromatic isocyanates and does not undergo the secondary cross-linking reactions commonly observed with aromatic isocyanates. The urethanes chosen for this study contained soft segments based on hydroxy-terminated polybutadiene (HTPBD) or polycaprolactone (PCL), which is a highly polar material in which extensive phase mixing should occur. In contrast, polyurethanes based on HTPBD should undergo more complete phase separation. Prepolymers were prepared with both of these soft segments and with available isocyanate contents varying from 3.7% to 5.7%. Diethyltoluene diamine (DETDA) was used to cure these prepolymers over a range of cure stoichiometries. Thermal transitions attributed to the hard and soft segments were observed in these urethanes using a Perkin-Elmer DSC-4. Dynamic mechanical behavior was measured using both a Polymer Labs dynamic mechanical thermal analyzer (DMTA) and a resonance apparatus developed at the NRL-USRD. [Sponsored by ONT.]

2:30

TT4. Bulk modulus thermoviscoelasticity theory for rubbery elastomers. J. Burns (Florida Institute of Technology, Melbourne, FL 32903) and P. S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

Earlier attempts to develop a free-volume theory of the dynamic bulk modulus of rubbery elastomers [J. Burns *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 79, S62 (1986)] led to separate expressions for the frequency dependence of the modulus at constant temperature and for the temperature dependence at constant frequency. These have now been combined to give a free-volume-based theory for the bulk modulus as a function of both frequency and temperature. For certain ranges of parameters that may

occur in practice, the theory does not permit master curves to be constructed by the usual WLF frequency shift method. Implications of this fact for dealing with experimental bulk modulus data are discussed.

2:45

TT5. An algorithm for applying the Williams, Landel, and Ferry method of reduced variables. James J. Dlubac, Richard J. Deigan, and Wayne T. Reader (Code 1905.2, Ship Acoustics Department, David Taylor Naval Ship R & D Center, Bethesda, MD 20084-5000)

The measurement of the dynamic complex modulus of viscoelastic materials on any apparatus is confined to a limited frequency and temperature range. Williams, Landel, and Ferry (WLF) [J. D. Ferry, *Viscoelastic Properties of Polymers* (Wiley, New York, 1980), 3rd ed. Chap. 11] reasoned that an increase in temperature produces an identical effect in a viscoelastic material as a decrease in frequency and vice versa. Empirically, then, there is a correspondence between temperature and frequency. The WLF method of reduced variables was proposed as a method of extending the effective frequency range of the dynamic modulus and loss factor. This technique involves measuring the complex modulus over a limited frequency range and a variety of temperatures. The data at different temperatures are then plotted versus frequency and shifted along the frequency axis until a smooth curve results. The effects are shifted modulus curves describing the complex modulus over a broad frequency range at some fixed temperature. Typically, the WLF shift requires several runs at temperatures close enough so that the modulus of one run overlaps the next. In this paper, the WLF method of reduced variables is reviewed and an algorithm for accomplishing the WLF shift is investigated. This algorithm shifts dynamic modulus data at various temperatures, the consecutive runs of which may not contain overlapping moduli.

3:00

TT6. Effects of aging on dynamic bulk moduli of several elastomers. J. Burns (Florida Institute of Technology, Melbourne, FL 32901), P. S. Dubbelday, and R. Y. Ting (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

Samples of natural rubber, butyl rubber, neoprene, and polyurethane were aged for a total of 24 months at room temperature in seawater. The dynamic bulk modulus was measured for each sample at intervals during the 2-yr aging period. Age-induced changes in the dynamic bulk moduli were relatively small in all cases, but there were distinct differences among the various types of elastomers tested. In general, aging effects appear considerably more pronounced for shear and Young's moduli than for bulk moduli. Possible reasons for this are discussed.

3:15

TT7. An automated system for characterizing the vibration damping properties of materials. M. L. Drake (University of Dayton research Institute, 300 College Park, Dayton, OH 45469), R. G. Smiley (Entek Scientific Corporation, Cincinnati, OH), and D. M. Hopkins (University of Dayton Research Institute, 300 College Park, Dayton, OH 45469)

In 1980 ASTM finalized the damping characterization standard E-756-80 around the resonant beam test method (M. L. Drake and G. E. Terborg, Technical Report AFWAL-TR-80-4093, 15 January 1976-31 December 1979). The University of Dayton, in conjunction with Entek Scientific, developed a computer-aided test system that automated the test procedures and data reduction equation prescribed in E-756. This software package, combined with the appropriate hardware system, results in an efficient workstation for conducting damping characterization studies of materials and designing effective damping systems utilizing the damping properties determined. The presentation will outline the philosophy and operation of the system, including discussions on the system auton-

omy and the function of each system component; a typical test setup, data reduction, and nomogram generation; and an example of a damping design development.

3:30

TT8. Dynamic material property measurements using the Polymer Laboratories' dynamic mechanical thermal analyzer. Richard J. Deigan, Wayne T. Reader, and James J. Dlubac (Code 1905.2, Ship Acoustics Department, David Taylor Naval Ship R & D Center, Bethesda, MD 20084-5000)

The basic principle of measurement of any dynamic mechanical property apparatus is to infer the elastic and loss moduli from the measured response of a sample and a known theoretical solution. The extent to which the measured response is modeled by the theoretical solution determines the validity of the inferred material properties. The purpose of this paper is to investigate the measurement of the dynamic moduli of viscoelastic materials using the dynamic mechanical thermal analyzer manufactured by Polymer Laboratories, Inc. The complex Young's modulus [$E^* = E'(1 + i\eta_E)$] or shear modulus [$G^* = G'(1 + i\eta_G)$] is determined at frequencies ranging from 0.01 to 200 Hz at temperatures ranging from -150°C to 300°C . The Young's modulus is found through the bending of either a double or single cantilever beam. The WLF shift is employed on data at several different temperatures to generate shifted modulus curves of E' and η_E and/or G' and η_G versus extended frequency for fixed temperature. Computer-generated shifted modulus curves of viscoelastic materials are discussed.

3:45

TT9. Viscoelastic constants for pressure release materials. Gary Caille, Jacek Jarzynski, Peter Rogers, and George W. Woodruff (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A comparison of experimentally and theoretically determined viscoelastic constants for closed cell neoprene rubber and corprene is presented. Both of these materials have air concentrations of greater than 50% by volume and exhibit large acoustic attenuation. The longitudinal sound speed is approximately 150 m/s for the closed cell neoprene rubber and 340 m/s for the corprene. The experimental constants determined are the complex Young's modulus and complex plane-wave modulus in the frequency range of 500 to 5000 Hz. Standard experimental techniques for measurement of these moduli are used. The experimental data are corrected for multiple internal reflections by use of the complex cepstrum signal processing technique. The theoretical predictions are based on multiple scattering models and quasistatic mean elastic field models. [Work supported, in part, by ONR Code 1125A.]

4:00

TT10. Dynamic shear modulus measurements of fiber reinforced polymers using forced torsional vibrations. Wayne T. Reader, James J. Dlubac, and Richard J. Deigan (Code 1905.2, Ship Acoustics Department, David Taylor Naval Ship R & D Center, Bethesda, MD 20084)

A forced torsional vibration system developed by the National Bureau of Standards [E. B. Magrab, J. Res. Natl. Bur. Stand. **89**, 193-207 (1984)] for the David Taylor Naval Ship Research and Development Center is used to measure the dynamic shear modulus of viscoelastic materials. The complex shear modulus $G^* = G'(1 + i\eta_G)$ is determined by measuring the relative amplitude and phase at the ends of a harmonically torqued right circular cylinder. The shear storage modulus and loss factor are found at frequencies ranging from 50 to 1000 Hz for fixed temperatures ranging from -40°C to 70°C . An algorithm based upon the Williams-Landel-Ferry method of reduced variables [J. J. Dlubac, *et al.*, J. Acoust. Soc. Am. Suppl. **1 82** (1987), preceding abstract] is applied to the isothermal data sets to generate curves of G' and η_G versus extended frequency range for a fixed temperature. A major feature of the system is that the cylindrical specimen dimensions may range up to 10 cm in diameter and 15 cm in height. The advantage, of course, is that the effective complex moduli of inhomogeneous materials may be determined. For

example, the system may be used to investigate the dynamic properties of viscoelastic polymers containing finite sized inclusions such as chopped fibers or platelet fillers. Examples will be presented of the system's use to examine the reinforcement provided by glass fibers of several lengths randomly dispersed throughout a viscoelastic polymer.

4:15

TT11. Constrained viscoelastic layer as used to dampen structures under multiple straining modes. M. Bonnet and B. Garnier (MetraVib R.D.S., 64 Chemin des Mouilles, 69130 Ecully, France)

The design of viscoelastic damping mechanisms requires both a thorough knowledge of the viscoelastic properties of materials and proper dimensioning and tools capable of handling all cases. This is particularly important when dealing with aerospace structures where the weight-to-performance ratio is critical. This point is illustrated using a specific example. Here, an aluminum honeycomb/carbon skin sandwich is damped to handle explosion shock waves over a frequency bandwidth of 1 to 10 kHz at a temperature of -20°C . Dynamic mechanical analysis (in modulus and loss angle) on a variety of materials led to the selection of an appropriate polyurethane elastomer. The resulting sandwich structure accounts not only for bending modes (using Ruzicka's theory), but for tension compression and shear modes as well (inspired from Y. V. K. Sadasiva Rao and B. C. Nakra). This compromise is obtained through the development of specific software, which is both simpler and quicker than a finite element analysis. A good agreement between theory and experiment at room temperature is shown. The waterfall visualization technique used is particularly useful in detailing apparent mechanisms for all configurations: bending, shearing, and tension compression. These results are so encouraging that further development of this technique is justified.

4:30

TT12. Measurement of complex electromechanical properties of highly damped piezoelectric materials. F. Douglass Shields (Physics Department, University of Mississippi, University, MS 38677) and Kurt M. Rittenmyer (Underwater Sound Reference Detachment, P. O. Box 568337, Orlando, FL 32856-8337)

The mechanical resonance technique for determining the dielectric, elastic, and electromechanical properties of piezoelectric materials has been developed and refined for over 40 years, as described in the different versions of the IRE-IEEE standard on piezoelectricity. The conventional method assumes the piezoelectric resonator is nearly mechanically lossless. This assumption is valid for some piezoelectric ceramic compositions, but is inaccurate for more highly damped materials used in hydrophone applications. New methods based on the measurement of the real and imaginary parts of the complex admittance have been devised by Smits [IEEE Trans. Sonics Ultrason. **SU-23**, 393-402 (1976)] and Saitah *et al.* [IEEE Ultrason. Symp. Proc. 620-623 (1985)]. One of these methods has been accepted and the parameter fitting routines improved upon to determine the electromechanical properties of several important piezoelectric materials that have a high degree of damping. The importance, applications, and limitations of this method will be described for these piezoelectric materials.

4:45

TT13. Surface waves in viscoelastic fluid. H. W. Jones, H. W. Kwan (Department of Engineering Physics, Technical University of Nova Scotia, Halifax, Nova Scotia B3J 2X4, Canada), and E. Yeatman (Department of Electrical Engineering, Imperial College, London, United Kingdom)

Viscoelastic fluids will, in principle, propagate Lamb and Rayleigh waves. In these circumstances, the shear modulus is frequency dependent in both its real and imaginary parts; therefore, it might be expected that some unusual effects will occur. In this paper, the propagation of such waves by reference to the viscoelastic properties of crude oil is described. Numerical calculations showing the properties of these waves under a range of conditions are presented.

Session UU. Underwater Acoustics VII: Propagation Modeling

Henrik Schmidt, Chairman

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

Chairman's Introduction—1:30

Contributed Papers

1:35

UU1. Numerical modeling techniques in underwater acoustics. Paul C. Etter (ODSI Defense Systems, Inc., 6110 Executive Boulevard, Rockville, MD 20852)

A summary of available underwater acoustic modeling techniques is presented, emphasizing those computer-implemented codes applicable principally to research in support of naval sonar design and operation. This comprehensive summary addresses three major classes of models: propagation loss, noise, and reverberation. Fifty propagation loss models are categorized according to five theoretical derivatives of the wave equation: ray theory, multipath expansion, normal mode, fast field, and parabolic approximation; further distinctions are made between range-independent and range-dependent solutions. Fourteen noise models are divided into the categories of ambient noise and beam-noise statistics; models in the latter category are further segregated according to analytic and simulation approaches. Fourteen reverberation/active sonar models are jointly described in addition to two multiple-component model operating systems; the reverberation models are distinguished according to cell and point scattering formulations. Supporting open-literature citations are provided to identify principal investigators and institutions. This summary affords a convenient baseline against which to define future developmental efforts in such areas as hybrid, 3-D, broadband, stochastic, shallow-water, and under-ice acoustic modeling.

1:50

UU2. Derivation of higher-order parabolic equations which include density variations. George H. Knightly and Donald F. St. Mary (Center for Applied Mathematics and Mathematical Computation, Department of Mathematics, University of Massachusetts, Amherst, MA 01003)

Several approaches to the derivation of parabolic equations are explored, which include general density functions, starting from the acoustic wave equation in two and three dimensions. In particular, higher-order (wide angle) parabolic equations are presented. Also presented are finite-difference discretizations of the equations that (1) attempt to alleviate the need for special consideration of the "continuity conditions" when one fluid medium interfaces with another, and (2) specifically take into consideration such interfaces. Mathematical comparisons are made between these two approaches.

2:05

UU3. Transverse cusp caustics produced by reflection and transmission: The caustic surface and optical simulations. Philip L. Marston and Carl K. Frederickson (Department of Physics, Washington State University, Pullman, WA 99164)

High-frequency sound reflected from curved surfaces or refracted by inhomogeneities may produce cusped caustics that open up roughly transverse to the direction of propagation. Though the better-known longitudinal cusp or arête is produced by a concave cylindrical wave front, transverse cusps are produced by distinctly noncylindrical wave fronts of the form $W(x,y) = -(a_1x^2 + a_2y^2x + a_3y^2)$, where $a_2 \neq 0$ and the distance z to the observation plane $\neq (-2a_1)^{-1}$ [Marston, J. Acoust. Soc. Am. **81**, 226-232 (1987)]. The wave front propagates in the z direction and in

the orthogonal (u,v) observation plane the caustic has the form $D(u - u_c)^3 = v^2$. The present research is concerned with the caustic surface generated by considering the z dependence of the caustic parameters D and u_c . Reflection and transmission problems that lead to this $W(x,y)$ in the paraxial approximation are discussed as well as a novel property of merging rays. Sound radiated by a point source so as to reflect from a surface whose height relative to the xy plane is of the form $h_1x^2 + h_2y^2x + h_3y^2$ produces a transverse cusp. The reflected wavefield is described by the Pearcey function. Transverse cusps were simulated by reflecting light from a surface of this form. [Work supported by ONR.]

2:20

UU4. Results of a very-low-frequency acoustic propagation experiment in the Cascadia Basin. Hassan B. Ali, Craig Fisher, Mona Authement (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004), and Jeff Beckleheimer (ODSI Defense Systems, Inc., 6110 Executive Boulevard, Rockville, MD 20852)

A series of very-low-frequency (VLF) measurements were conducted recently by the Naval Ocean Research and Development Activity (NORDA) in the Cascadia Basin off the Oregon coast. Using a fixed vertical array of 16 hydrophones and a distribution of ocean bottom seismometers (OBS), the responses to cw, explosive, and air gun sound sources were monitored. The partitioning of acoustic energy between the waterborne and bottom paths is examined, using the relative responses of OBS and water-column hydrophones. Comparisons between the experimental results and predictions based on standard numerical models of acoustic propagation are used to ascertain the effects on VLF propagation of bottom shear and range-dependent topography.

2:35

UU5. High-speed normal mode calculations via Milne's equation. F. J. Ryan (Code 541, Naval Ocean Systems Center, San Diego, CA 92152-5000)

A major impediment to the use of normal mode methods in deep water and/or at high frequencies is the calculational time per mode. A fast, numerically stable method of mode calculation is described which is based upon Milne's numerical solution of the Schrödinger equation. The gist of Milne's method is a phase-amplitude transformation of the linear second-order ODE for the modal depth functions into a nonlinear ODE for the quantum momentum. The resulting nonlinear equation has improved numerical stability, particularly for high-order modes. In the limit of large wavenumber, the method reduces to the conventional JWKB form, but, unlike JWKB, it is valid at turning points. An accurate and efficient computational method is developed that yields second-order convergence to an eigenvalue. The technique is demonstrated for single and multiple duct sound-speed profiles and compared to conventional propagator matrix approaches for modal calculations.

2:50

UU6. An efficient enhancement of finite-difference implementations for solving parabolic equations. J. S. Robertson, D. C. Arney (Department of Mathematics, U.S.M.A., West Point, NY 10996-1786), W.

The parabolic approximation method is widely recognized as useful for accurately analyzing sound transmissions in diverse ocean environments. One reason for its attractiveness is because solutions are marched in range, thereby avoiding the massive internal storage required when using the full wave equation. Present implementations employ a range step size that is prescribed either by the user or by the code and remains fixed for the duration of the computation. An algorithm is presented in which the range step is adaptively selected by a procedure within the implicit finite-difference (IFD) implementation of the parabolic approximation. An error indicator is computed at each range step, and its value is compared to a user-specified error tolerance window. If the error indicator falls outside this window, a new range step size is computed and used until the error indicator again leaves the tolerance window. For a given tolerance, the algorithm generates a range step size that is optimal in a specified sense and which often leads to large decreases in run time. Additional related modifications to the IFD implementation will also be discussed. Several examples are presented that illustrate the efficacy of the enhanced algorithm. [Work supported by ONR.]

3:05

UU7. Effects of phase errors on pulse synthesis. David H. Wood (Code 3332, Naval Underwater Systems Center, New London Laboratory, New London CT 06320) and Robert P. Gilbert (Center for Scientific Computation, University of Delaware, Newark, DE 19716)

One way of modeling pulse propagation in the ocean is to decompose the pulse into a weighted sum of functions $e^{i\omega t}$, summed over various values of ω . One then solves the Helmholtz equation numerically for each required value of ω . The weighted sum of these fields then gives the desired response to the pulse in question. An alternate approach solves the parabolic equation for each required value of ω , rather than the Helmholtz equation, thereby introducing phase errors. These two results can be related by an integral transform, or *transmutation*, that converts the approximate solution obtained from the parabolic equation into the true solution obtained from the Helmholtz equation. As a result, corrections can be made and/or errors can be estimated or bounded.

3:20

UU8. Rigorous Gaussian-beam modeling of source functions for acoustic radiation and propagation. John J. Maciel (Missile Systems Division/Radome Section, Raytheon Co., Bedford, MA 01730) and Leopold B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, NY 11735)

Gaussian beam functions have favorable properties for propagating acoustic wavefields through complicated ocean environments. Use of these functions in propagation algorithms requires a decomposition of the actual or induced source distribution into a superposition of Gaussian beams. In implementations of the "Gaussian beam method" so far, the decomposition is not unique because of freely assignable parameters in the beam stack. This causes difficulties with *a priori* predictability [Lu *et al.*, *Geophys. J. R. Astron. Soc.* (1987); Niver *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S9 (1987)]. The source representation problem can be addressed rigorously by performing the decomposition on a lattice in a discretized (configuration-spatial wavenumber) phase space. The formulation of this discretization scheme [Bastiaans, *Opt. Eng.* **20**, 594 (1981)] is reviewed and then applied to radiation from a cosine aperture test field distribution. It is shown how successive addition of individual displaced and (or) rotated beams with narrow, wide, or "matched" waists systematically homes in on the independently calculated reference solution, although each selection strategy emphasizes different regions in the phase space. The utility of the various options is discussed, as are the implications for synthesis of aperture fields from measured farfield data.

3:35

UU9. A full wave solution for propagation in horizontally stratified elastic media with full forward and backscatter. Ziad Haddad (AT&T Bell Laboratories, 14A420, Whippany Road, Whippany, NJ 07981)

A numerically efficient procedure for finding complete solutions to wave propagation problems in elastic media whose parameters depend on depth only, but where one of the interfaces (e.g., water/air) is allowed to vary (under some restrictions) with range, is derived from the method recently proposed by H. Schmidt and F. Jensen. This extension applies to the case where the depth at an interface expressed as a function of range is a sum of sinusoids. The roughness imposes energy coupling equations that must be satisfied by sets of wavenumber components of the pressure field, grouped according to the wavelengths of the sinusoids. The importance of the coupling depends on the amplitudes of the sinusoids, and accounts for all nonevanescing forward and backscattering. Plane-wave reflection coefficient calculations using this method agree with exact computations based on work by R. Holford [*J. Acoust. Soc. Am.* **70**, 1116-1128 (1981)]. This procedure is further extended to allow the rough interface, say, the free surface, to move (under some restriction) with time, thus allowing a rapid complete solution of the wave equation including full Doppler effects.

3:50

UU10. Rough surface scattering using the parabolic wave equation. Eric I. Thorsos (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195)

The accuracy of the parabolic wave equation (PE) for calculating scattering from randomly rough, pressure-release surfaces has been examined through comparison with exact numerical results based on solving an integral equation. To facilitate comparison with exact results, the PE problem is also converted to one of solving an integral equation: the "PE integral equation." For the cases examined to date, the PE accurately predicts low grazing angle ($\theta_g < 20^\circ$) forward scattering, including multiple scattering and shadowing effects. As the grazing angle increases, it is found that the main error is an angular shift arising from propagation to and from the surface, not from the surface scattering itself. Finally, a method has been devised to compute low grazing angle backscattering with the PE. This requires the sequential solution of two integral equations to account for multiple scattering in both the forward and backward directions. [Work supported by ONR.]

4:05

UU11. Multiple array processors for three-dimensional acoustic problems. Yu-chiung Teng (Aldridge Laboratory of Applied Geophysics, Columbia University, New York, NY 10027)

Presently available analytic techniques for solving wave propagation problems are only useful for simple cases. In realistic cases, the presence of inhomogeneities and irregular boundary conditions defies analytical solutions. One of the best numerical techniques suitable for solving wave propagation problems in a complex geological medium is the finite element method. In principle, the finite element method is capable of dealing with any two- or three-dimensional acoustic and elastodynamic problems. However, the computer core storage and computing time for solutions increase greatly with each dimension and are beyond the capabilities of the conventional computers. It is, therefore, always desirable to search for alternatives that may reduce the computational labor. In taking advantage of the current advances in hardware technology, the goal is to develop the finite element algorithm for an extraordinarily fast and inexpensive computer system. In particular, the implementation of the finite element method on multi-array processor system was considered. The algorithm, based on the nodal-podal-point-oriented approach, for implementing the finite element method on the parallel computation system has been developed. As a demonstrative example, the three-dimensional problem for the acoustic case has been successfully tested on the IBM loosely Coupled Array Processors (ICAP) by using one to eight processors. By using the $31 \times 31 \times 31$ -element model on PRIME 750, the computing time for 200 time steps is about 8.8 h and I/O time is 1.7 h. If using the ICAP1 to calculate the same model, the elapse time is about 16.7 min with two processors, 8.8 min with four processors, and 4.9 min with eight processors.

Session VV. Underwater Acoustics VIII: Volume and Object Scattering

C. P. de Moustier, Chairman

Scripps Institution of Oceanography, Marine Physical Laboratory A-005, La Jolla, California 92093

Chairman's Introduction—1:30

Contributed Papers

1:35

VV1. Average intensity of an acoustic beam after propagating through a turbulent ocean. J. H. Tarnag and C. C. Yang (CSSL, Department of Electrical Engineering, The Pennsylvania State University, University Park, PA 16802)

The average intensity of an acoustic beam propagating in a turbulent ocean is evaluated using the path integral technique. Because of the existence of the vertical sound-speed profile in the ocean, there is more than one stationary path for the path integral formulation. Also, these stationary paths are curved rays which are derived by the method of ray tracing. The effects of random fluctuations in sound speed are studied by introducing both internal waves and temperature fine structure. The results show that the acoustic beam becomes broader and asymmetric along the vertical axis. The effects of the broadening, asymmetry, and multiple stationary paths are examined in detail. Interference among multiple stationary paths is expected. In addition, these results are compared with those of others obtained by using different approaches. [Work supported by ARL, The Pennsylvania State University.]

1:50

VV2. Acoustic pulse propagation in a turbulent ocean. J. H. Tarnag and C. C. Yang (CSSL, Department of Electrical Engineering, The Pennsylvania State University, University Park, PA 16802)

Temporal behavior of acoustic pulse propagation in a turbulent ocean is investigated by using the method of temporal moments. In this method, the zeroth, first, and second temporal moments are related to the total energy of the pulse, arrival time, and pulse width, respectively. To compute these moments, instead of the two-frequency mutual coherence function, only the coefficients in a power series expansion of this function are required. In particular, only the coefficients up to the second-order terms are needed. The path integral technique, incorporated with the method of ray tracing, is used for evaluating those coefficients. The vertical sound-speed profile is taken into account. The turbulences in the ocean include the internal waves and temperature fine structure. For mathematical simplicity, a bilinear sound-speed profile is considered, and it is assumed that the propagation range along a horizontal axis is short, such that the rays always stay either above or below the depth of minimum sound speed. More general results can be extended easily. [Research supported by ONR.]

2:05

VV3. Sensitivity of acoustic fields to environmental uncertainties. L. B. Dozier, H. Freese, E. S. Holmes, and F. D. Tappert (Science Applications International Corp., 1710 Goodridge Drive, McLean, VA 22102)

A PE model has been used to compare 2D cw acoustic fields $p(x,z)$ propagating in sound-speed fields $c(x,z)$ to a reference acoustic field $p_r(x,z)$ propagating in the reference sound-speed field $c_r(x,z)$. Samples of the range- and depth-dependent sound-speed difference fields, $\delta c(x,z) = c(x,z) - c_r(x,z)$, were generated from physical models of (1) instrumental errors, (2) internal waves, and (3) baroclinic waves (mesoscale). The square modulus of the normalized inner product over depth,

called $\rho(x)$, with $\rho = 1$ corresponding to $\delta c = 0$, was used as a measure of the distance between p and p_r . Results are presented as curves of $\rho(x)$ in dB units for various frequencies, depending on the magnitude of δc . It was found that $\rho(x)$ decreases monotonically with range, on the average, and saturates in range at a value that decreases with increasing frequency at the rate of about 3 dB/oct. The range to saturation, called the "predictability horizon," is surprisingly short for most frequencies of interest for moderate δc .

2:20

VV4. Modeling of acoustic transmission in the Straits of Florida. Charles L. Monjo and H. A. DeFerrari (Department of Applied Marine Physics, RSMAS, University of Miami, Miami, FL 33149)

Acoustic transmission experiments, over a 24-km range in the Straits of Florida, have been modeled using ray theory. Three acoustic moorings and three thermister moorings were operated for a 30-day period. Pulse responses (460-Hz carrier with 100-Hz bandwidth) were recorded every 12 min, and temperature at several depths every half hour. Pulses were found to be highly variable in travel time, duration, and shape. The MEDUSA ray tracing eigenray model was used to explain the pulse formation and variation, using the thermister time series as input. Model predictions of pulse responses agree with the data. Positive temperature anomalies bring about long duration pulses, 200 ms, while negative temperature anomalies bring about short duration pulses, 50 ms. The mechanism behind the pulse duration variation was found to be a function of bottom loss, not a focusing effect of the sound-speed profile. Negative temperature anomalies tend to direct grazing eigenrays more steeply into the bottom, causing greater bottom loss to the front end of the pulse. The front end of the 200-ms pulse is "lost" below the noise level and a short duration 50-ms pulse results. [Work supported by ONR.]

2:35

VV5. Acoustic monitoring and plume mapping of drilling fluid discharge. John J. Tsai and John R. Proni (NOAA/AOML, Ocean Acoustic Division, Miami, FL 33149)

An approach to monitor drilling fluid discharge from an oil rig and to map its plume distribution by high-frequency acoustics was proposed. Experiments were conducted at two nearby rigs of the East Flower Garden Bank in the Gulf of Mexico, and data were used to provide vertical plume structure and two-dimensional distribution maps at fixed depth. The ability to monitor the discharged plume continuously in time and space makes the acoustic technique more convenient and cost effective than other conventional methods.

2:50

VV6. The influence of bubbles on sea surface backscatter measurements. Paul D. Koenigs, Joseph M. Monti (Naval Underwater Systems Center, New London, CT 06320), Bernd Nuetzel, and Heinz Herwig (Forschungsanstalt der Bundeswehr fuer Wasserschall- und Geophysik, 23 Kiel 14, West Germany)

The results from a recent sea surface acoustic scattering experiment,

which was conducted in the North Sea, are presented with accompanying sea surface roughness parameters and subsurface bubble information. The acoustic data were obtained utilizing a high-resolution (narrow beam-width) pulsed parametric sonar transmitter and conventional receiver. Scattering strength values were obtained as functions of frequency (3–18 kHz) for wind speeds from 2–45 knots. It appears that the backscattering strength at 30° grazing angle is caused by the high-frequency wavenumber spectrum at low wind speeds and by subsurface bubbles at high wind speeds. The backscattering strength shows strong fluctuations in the intermediate region caused by both scattering mechanisms.

3:05

VV7. Nearfield calculations from elongated objects and a comparison with the farfield form function. M. F. Werby, Guy Norton (NORDA Numerical Modeling Div. NSTL, MS 39529), and G. Gaunaurd (NSWS, White Oak, Silver Spring, MD 20910)

It is usual to examine farfield form functions when examining scattering from submerged objects. In this study, what happens when one examines the bistatic angular distributions in the very nearfield for elongated spheroids was determined. The distance from the field is then progressively moved from the object in multiples of the semimajor axis until the results agree with the farfield form function. It is found that this does not occur until one is a distance greater than 20 times the largest diameter from the object. When one is close to the object, the calculations reflect the fact that a shadow is cast on the observer and, therefore, the pattern is rather broad close up and it narrows as one progressively moves from the object. This last effect can be obtained using a simple geometrical expression that agrees with the theoretical calculations.

3:20

VV8. Eigenvalue methods in the solution of acoustical scattering problems from submerged objects. Guy Norton, Richard Keiffer, M. F. Werby (NORDA Numerical Modeling Div. NSTL, MS 39529), and L. H. Green (Martin Marietta Baltimore, Baltimore, MD 21220)

Waterman's extended boundary condition (EBC) method has proven to be extremely successful in dealing with scattering from submerged targets. An alternate and useful computational method (which is related to Enskog's method) separately considers the Helmholtz integral equation at exterior and interior points, as does the EBC method. However, the interior problem can be shown to transform to an eigenvalue problem, which produces eigenstates that span the space of the displacement on the object surface. The eigenstates can be used directly to solve the exterior problem and yield a numerically stable and convergent solution. This new method avoids problems encountered by the EBC method. For example, in the usual EBC approach, the unknown surface terms are expanded on a known, but to some extent arbitrary, basis set, with unknown expansion coefficients. The number of expansion terms on the surface and those of the incident partial waves must match. Computationally, this requirement can often dictate there be many more incident partial waves than strictly required for convergence of the incident field. This leads to small-

incidence high-order components, that, in turn, render the resulting matrix problem ill-conditioned. The method used in this study is presented with several representative numerical examples including objects of aspect ratios of 30 to 1 for $kL/2$ values to 120.

3:35

VV9. Scattering from an ellipsoid submerged in the ocean. Michael D. Collins (Department of Engineering Sciences and Applied Mathematics, Northwestern University, Evanston, IL 60201) and Michael F. Werby (Naval Ocean Research and Development Activity, NSTL Station, MS 39529)

A time periodic source and an ellipsoid are submerged in a stratified ocean several kilometers apart. The method of matched asymptotics is used to derive an approximate expression for the field scattered from the ellipsoid. The matched asymptotics solution consists of an inner solution valid near the scatterer and an outer solution valid away from the scatterer. The two parts of the solution are matched in the region in which they are both valid. The resulting solution is composed of solutions to problems existing models can handle. The asymptotic limit applied to arrive at the result requires that the scatterer is small relative to the depth of the ocean and is away from the ocean surface and bottom. However, the method should be sufficiently robust to allow these assumptions to be relaxed greatly. This method is only weakly dependent on frequency; the frequency must be on the order of 10 Hz or greater. [Work supported by ONR and NORDA.]

3:50

VV10. Limitations of mechanical structures as wave vector filters. Y. F. Hwang and G. Maidanik (David Taylor Naval Ship R&D Center, Bethesda, MD 20084)

Although the high-wavenumber content in a highly subsonic (low Mach number) turbulent boundary layer (TBL) is known with reasonable certainty, the low-wavenumber content is not. In a wave vector filter chosen and designed to measure the low-wavenumber content of TBL, the major sensitivity region is appropriately placed in that spectral domain. Inevitably, some minor sensitivity regions lie in the high- (convective) wavenumber region. These minor sensitivity regions are then fed by the spectral pressure ridge in the TBL. This feeding may substantially contaminate the measurements. The proper choice and the novel design of a filter is judged, by and large, by its ability to subdue such contamination. In this paper wave vector filters, for measurements of the low-wavenumber content of TBL, as provided by ideal rectangular and circular panels, are compared and contrasted. In this examination, the contamination caused by the high-wavenumber content in the minor sensitivity regions is estimated by simulating it as a low-wavenumber content in the major sensitivity region. These estimated equivalent low-wavenumber contents constitute the lower bounds that the wave vector filters are capable of measuring.

Session WW. Physical Acoustics VIII: Acoustic Levitation

Anthony A. Atchley, Chairman

Department of Physics, Code 61AY, Naval Postgraduate School, Monterey, California 93943

Chairman's Introduction—8:25

Contributed Papers

8:30

WW1. Acoustic levitation at high temperatures in the microgravity of space. Charles A. Rey, Dennis R. Merkley, Gregory R. Hammarlund, and Thomas J. Danley (Intersonics, Incorporated, 3453 Commercial Avenue, Northbrook, IL 60062)

Acoustic levitation of a small specimen is obtained in the energy well produced by a single axis arrangement consisting of a sound source and a small acoustic reflector. At high temperatures, the acoustic forces are generally insufficient to levitate a specimen except in the microgravity environment available in space. A single axis acoustic levitator (SAAL) has been built to levitate a specimen inside a high-temperature furnace where it may be heated, melted, cooled, and solidified while being positioned without physical contact. The acoustic field configuration in such a containerless processing device has been analyzed and the expected levitation or positioning forces are calculated for the specific case of the NASA-SAAL experimental hardware as flown on the Space Shuttle on the STS-61A mission. This experiment successfully levitated and processed three samples at temperatures from 600 °C to 1550 °C. Experimental data are presented and the results compared with those predicted. [Work supported by NASA.]

8:45

WW2. Acoustic levitation in the presence of gravity. M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109) and P. Collas (Department of Physics and Astronomy, California State University, Northridge, Northridge, CA 91330)

A theory of acoustic levitation in the presence of gravity has been developed for arbitrarily oriented chambers of rectangular, cylindrical, and spherical geometries. Simultaneous mode excitation was also included in the analysis. Examples of the total potential and force profile will be given for commonly used levitation modes as a function of chamber orientation. Additional levitation positions are obtained within the chamber for appropriately chosen modes and orientations. Levitation positions within a chamber where the acoustic forces are isotropic will be presented for the case of small kR . The effects of mode degeneracy on the total potential and force for the case of the triple axis levitator flown on the Space Shuttle will also be discussed. [Work supported by NASA.]

9:00

WW3. Shape of acoustically levitated samples in a resonant rectangular chamber. H. W. Jackson, M. Barmatz, and C. Shipley (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

A generalization of Laplace's formula applicable to vibrating surfaces was used to develop expressions for the equilibrium shape and position of drops and bubbles in a resonant rectangular chamber. Expanding various physical quantities in terms of spherical harmonics and applying the quantum-mechanical angular-momentum gradient formula leads to tractable expressions valid for large as well as small ka . New results for volume renormalization of drops and bubbles were also derived. The anal-

ysis takes into account acoustic radiation, surface tension, gravity, and compressibility effects. Salient features of this theory applicable to acoustic levitation will be presented. Calculated shape properties will be compared to existing results of one-dimensional theories. Also, the acoustic force dependence on ka , obtained from equilibrium position calculations, will be compared to Hasegawa's results. [Work supported by NASA.]

9:15

WW4. Boltzmann-Ehrenfest adiabatic principle applied to acoustic forces in a single-mode levitator. S. Putterman, Joseph Rudnick (Physics Department, University of California, Los Angeles, CA 90024), and M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

An application of the Boltzmann-Ehrenfest adiabatic principle yields a fundamental relationship between the acoustic potential, acting on a sample positioned in a single-mode cavity, and the shift in the resonance of an isolated mode. The theory is very general and applies to a sample and resonant cavity of arbitrary shape and dimension. In the case of a small sample in a lossless cavity, the strict proportionality between the potential and frequency shift follows from a fairly simple argument. One useful application of this relationship is the determination of positioning forces and torques on a levitated sample from frequency shift measurements. The results of experimental measurements of the equilibrium position and orientation of a levitated sample will be presented, and their consistency with theoretical predictions will be discussed. [Work supported by NASA.]

9:30

WW5. Theory of oscillational instabilities in acoustic levitators. Joseph Rudnick (Physics Department, University of California, Los Angeles, CA 90024) and M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

A general theory describes the oscillational instabilities of an acoustically levitated sample in a resonant cavity. The theory, based on a Green's function method, predicts the onset of instabilities and their saturation. Important factors controlling these instabilities include drive frequency, spatial structure of the levitation mode, and the size of the sample. The instability in a zero gravity environment has fundamentally different onset characteristics than in finite gravity. Calculations reveal hysteresis and saturation effects. Many of the present results are consistent with a previously developed empirical model of the instabilities [S. L. Garrett and M. Barmatz, *J. Acoust. Soc. Am. Suppl.* 1 77, S21 (1985)]. Typical examples of the above effects will be discussed. [Work supported by NASA.]

9:45

WW6. Eigenfrequencies of axisymmetric cavities: Numerical calculations. Arthur E. Woodling and James B. Mehl (Physics Department, University of Delaware, Newark, DE 19716)

The eigenfrequencies of axisymmetric, hard-walled acoustic cavity resonators have been calculated by using an integral-equation representa-

tion of the Helmholtz equation with Neumann boundary conditions. For axisymmetric shapes, the integral equation is one-dimensional. Numerical results have been obtained for a perfect sphere, and for slightly deformed spheres, and for cylinders with an internal sphere on axis. The results for slightly deformed spheres are in close agreement with the predictions of boundary-shape perturbation theory, from which they typically differ by less than 6 parts in 10^6 . The results for cylinders with internal spheres are in excellent agreement with measurements [M. Barmatz *et al.*, J. Acoust. Soc. Am. 73, 725-732 (1983)].

10:00

WW7. Perturbation calculations of eigenfrequencies of cylindrical cavity with internal spherical obstacle. James B. Mehl and Robert N. Hill (Physics Department, University of Delaware, Newark, DE 19716)

The eigenfrequencies of a hard-walled acoustic resonator are found by solving the Helmholtz equation with Neumann boundary conditions. A cavity C with an internal hard obstacle B of vanishing size was considered. As the size of the obstacle shrinks to zero, the eigenfunction does not approach an eigenfunction of the empty cavity. Thus a singular perturbation theory is required. A formalism was developed and tested by calculating the eigenfrequencies of a cylinder with an internal sphere on the axis. The results are compared with theoretical values determined by a boundary-integral-equation method (A. E. Woodling, Ph.D. thesis, University of Delaware, 1986) and experimentally [M. Barmatz *et al.*, J. Acoust. Soc. Am. 73, 725-732 (1983)]. [Work supported in part by NASA.]

10:15

WW8. Eigenfrequencies of a cylindrical cavity with an internal off-axis spherical obstacle. M. Barmatz, G. Aveni (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109), and James B. Mehl (Physics Department, University of Delaware, Newark, DE 19716)

A hard spherical obstacle is mounted inside a cylindrical cavity at a variable position. The resonance frequencies of the cavity were determined as a function of the position of the sphere for obstacle positions on and off the axis of the cylinder. The dependence of the resonance frequency on the sphere position is important in analyzing the stability of acoustically levitated samples. The results are in good agreement with theoretical calculations based on the singular perturbation theory of Mehl and Hill. The measurements will also be compared to a new theory that utilizes the Boltzmann-Ehrenfest adiabatic principle to predict the spatial dependence of the acoustic potential from the resonance frequencies. [Work supported by NASA.]

10:30

WW9. A new acoustic levitation device using the interference sound field generated from two opposed radiators. Charles A. Rey, Thomas J. Danley, Dennis R. Merkley, and Gregory R. Hammarlund (Intersonics, Incorporated, 3453 Commercial Avenue, Northbrook, IL 60062)

A new technique for acoustically levitating a specimen is described. The technique deviates from a previous single-axis acoustic levitation method that utilizes a single-acoustic radiator and a passive reflector to generate the interference produced sound field. The new method involves the use of two opposing focusing acoustic radiators that produce an inter-

ference field that has capabilities for the levitation and positioning of small objects or liquid drops. The theoretical predictions regarding this technique are examined and those results are compared with laboratory measurements. The nature of the apparatus and its capabilities is also addressed. [Work supported by NASA.]

10:45

WW10. Specimen translational control capabilities using an opposed radiator acoustic levitation system. Dennis R. Merkley, Charles A. Rey, Gregory R. Hammarlund, and Thomas J. Danley (Intersonics, Incorporated, 3453 Commercial Avenue, Northbrook, IL 60062)

Recent developments on an opposed sound-source single-axis acoustic levitation system are discussed. New and improved specimen positioning control capabilities are made available by parametric variations of the electronic drive signals. Various methods are described in which a specimen can be held in one position or translated along the axis of the levitation by simple electronic control signals. A technique is also described to provide increased specimen positioning stability using a closed loop system. [Work supported by NASA.]

11:00

WW11. Enhanced acoustic levitation using horn loaded and focusing radiators. Thomas J. Danley, Charles A. Rey, Dennis R. Merkley, and Gregory R. Hammarlund (Intersonics, Incorporated, 3453 Commercial Avenue, Northbrook, IL 60062)

An improved piezoelectric driven, focusing acoustic radiator for acoustic levitation has been designed, tested, and evaluated. This device displays a significant improvement in focusing to produce better specimen stability and energy well shaping as well as increased transducer power bandwidth and electroacoustic conversion efficiency. Corresponding effects resulting from the horn loading of this acoustic radiator are considered as well as the benefits of using the radiator in conjunction with hot wall high-temperature furnaces. In order to aid in the evaluation of these acoustic transducer improvements, a simple and useful technique for acoustic field mapping will also be discussed. [Work supported by NASA.]

11:15

WW12. Enhancements of acoustic levitation in the presence of increased gas pressure. Gregory R. Hammarlund, Dennis R. Merkley, Charles A. Rey, and Thomas J. Danley (Intersonics, Incorporated, 3453 Commercial Avenue, Northbrook, IL 60062)

Acoustic levitation for containerless materials processing has been investigated for use in the 1-g environment. Enhancements have been obtained by increasing the gas pressure of the acoustic levitation system. Interdependence of gas pressure, sound pressure level, acoustic positioning force, and gas temperature has been measured. Also, nonlinear acoustic effects and harmonic content within the acoustic field are examined as a function of pressure. Finally, results of 1-g acoustic levitation experiments utilizing increased gas pressures are presented. [Work supported by NASA.]

Session XX. Psychological and Physiological Acoustics VIII: Binaural Hearing

Leslie R. Bernstein, Chairman

Department of Psychology, Psychoacoustics Laboratory, University of Florida, Gainesville, Florida 32611

Contributed Papers

8:30

XX1. Ear advantages for monaural periodicity detection. James A. Bashford, Jr. and Richard M. Warren (Department of Psychology, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

Monaural asymmetries were found for periodicity detection in experiments using repeated 200-ms segments of Gaussian noise (repetition frequency 5 Hz). In experiment 1, an overall left ear advantage was found for repeated noise delivered monaurally and opposed by contralateral silence. In experiment 2, lateralization of the monaural signal was abolished by simultaneous presentation of on-line noise to the opposite ear (contralateral induction caused the monaural signal to be perceived as centered on the medial plane). Although this manipulation eliminated the possible influence of attentional biases favoring one of the sides, ear advantages were still obtained. Alternative mechanisms will be discussed, including possible asymmetries in active subcortical processing of periodicity information. [Work supported by AFOSR.]

8:45

XX2. Binaural envelope correlation detection as a function of frequency separation. Virginia M. Richards (Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32611)

Recent comodulation masking release (CMR) experiments suggest that listeners are able to determine whether the envelopes of dichotically presented bands of noise are temporally similar (no signal) or different (signal added to one of the two bands, of noise). In an effort to determine the sensitivity to changes in envelope synchrony as a function of the frequency separation of the two bands of noise, listeners discriminated between noise bands whose envelopes were either identical or statistically independent. The noise bands were 100 Hz wide. The band presented to the left ear was centered at 2625 Hz; the band presented to the right ear was centered at values between 2625 and 5250 Hz. Although there were considerable individual differences, discriminations tended to be best when both noise bands were centered at 2625 Hz. Increasing the frequency separation between the two bands of noise led to poorer performance, but the percentage of correct responses remained above 60. Subsequent experiments indicate that the discrimination is based on envelope similarity rather than the similarity of the power spectrum. [Research supported by NIH.]

9:00

XX3. Effects of forward masker fringe on binaural detection. B. D. Simpson and R. H. Gilkey (Signal Detection Laboratory, Central Institute for the Deaf, Saint Louis, MO 63110)

Yost [J. Acoust. Soc. Am. **78**, 901-907 (1985)] found that detectability of a 20-ms dichotic signal ($S\pi$) in a 20-ms diotic noise (No) was not affected by the presence of a 500-ms dichotic forward fringe ($N\pi$). However, Kollmeier and Gilkey [J. Acoust. Soc. Am. Suppl. **1** **74**, S39 (1982)] varied the onset time of a 20-ms $S\pi$ signal in a 750-ms noise that switched, after 375 ms, from $N\pi$ to No , and found that the $N\pi$ noise reduced detectability even when the signal was well into the No noise. They concluded that the $N\pi$ noise acted as a forward masker. The present study investigates detectability of a 500-Hz, 20-ms $S\pi$ signal in a 20-ms No

masker as a function of the signal onset time. The masker is preceded by quiet or an $N\pi$ forward fringe and followed by quiet, an No , or $N\pi$ backward fringe. Overall, the results are compatible with a forward masking interpretation. Further, when the signal onset is simultaneous with the onset of the 20-ms masker, the $N\pi$ forward fringe reduces detectability. [Work supported by NSF and AFOSR.]

9:15

XX4. Further research on free-field masking. R. Wayne Gatehouse (Department of Psychology, University of Guelph, Guelph, Ontario N1G 2W1, Canada)

Gatehouse (Proc. 12th Int. Congr. Acoust., B2-7, 1986) compared free-field masked and unmasked thresholds (two-alternative forced-choice procedure) with a WN masker fixed at 0° while 5 tonal probes were delivered from 11 frontal plane positions. Masking was not necessarily greatest for masker-signal coincidence but varied with frequency and signal position. For 1.0- and 2.5- kHz signals, detectability was best and approximately equal at 0° and $+90^\circ$, while for the others (0.25, 0.5, and 4.0 kHz) there were generally lower thresholds as signal and masker separated. The task was repeated now with 0.5-, 1.0-, and 2.0-kHz probes and the masker set at 0° and $+45^\circ$. Again, free-field masking varied with both signal position and frequency, and best and least detection was not solely determined by azimuthal separation of the masker and signal. On average, masking level differences were approximately 24.0 dB for 0.5-, 11.5 dB for 1.0-, and 13.5 dB for 2.0-kHz signals. The apparent differences between earphone-derived and free-field masked thresholds, which seem more subject to frequency and positional variations, will be discussed.

9:30

XX5. Resolution of pairs of simple and complex sounds in simulated auditory space. Pierre L. Divenyi and Susan K. Oliver (Speech and Hearing Research, V.A. Medical Center, Martinez, CA 94553)

A simulated acoustic replica of a sound source moving along a circle with a 4-m radius in the frontal horizontal half-plane was generated by obtaining, in 5-deg azimuthal steps, external ear transfer functions for both ears of an artificial head placed in an anechoic room. When an arbitrary sound is passed through the digital filter defined by the left and right transfer functions corresponding to a given angle, the sound will acquire a subjective azimuth comparable to which the transfer function was measured [J. Blauert and P. Laws, *Acustica* **29**, 273-277 (1973)]. This technique was used to measure resolution thresholds for pairs of amplitude-modulated as well as pure sinusoids. With the listener required to identify the relative location of the two sounds in a two-alternative forced-choice paradigm, in one set of experiments the two sounds were fixed and the minimum audible angular separation was obtained, whereas, in another set, the angular separation was fixed and the minimum modulation- or carrier-frequency separation was measured. Generally, both kinds of resolution are better around 0 azimuth than at the sides: In the "auditory fovea," simultaneous resolution for pure tones and AM sounds (a 3- or 6-kHz carrier sinusoidally modulated in the 150- to 400-Hz range) is almost as good as the jnd obtained in the usual sequential presentation. The poorest spatial resolution was obtained for a pair of 6-kHz carriers modulated

9:45

XX6. The enhancement of binaural source location by scuba divers. Stewart A. L. Glegg, John Kloske, Chris Hansen, and Terry R. Johnson (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

This paper will describe the development of a passive source localization device to enhance the ability of scuba divers to locate acoustic sources. The device consists of a hat that is worn by the scuba divers and provides a distinct null in the observed acoustic field when the diver is facing the acoustic source. Tests were carried out in a 25-ft-deep, 198-ft-diam tank at the EPCOT Center in Orlando, FL on 25 subjects. The results of these tests show that there is a clear improvement using this device, which is comfortable to wear and suitable for most applications.

10:15–10:30

Break

10:30

XX8. Effects of roving level on binaural detection and discrimination on and off midline. J. Koehnke and H. S. Colburn (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139 and Boston University, 48 Cummington Street, Boston, MA 02215)

NoS π detection thresholds and interaural time and intensity jnd's were measured both with and without a 10-dB random rove applied to the overall level. The stimulus for discrimination and the masker for detection were a 1/3-oct noise band centered at 500 Hz; for detection, the target was a 500-Hz tone burst. Thresholds and jnd's were measured using an adaptive, four-interval, two-alternative forced-choice procedure for several interaural time and intensity reference conditions including differences of 0 and 24 dB and 0 and 600 μ s. Results indicate no effect of roving level for the 0-dB, 0- μ s reference condition for binaural detection or interaural discrimination. For NoS π detection, there is no effect of roving level or reference condition on thresholds. For interaural time discrimination, jnd's are larger off midline and roving level variation has no consistent effect on performance. For interaural intensity discrimination, jnd's generally increase off midline and are usually larger when measured with roving level than without. [Work supported by NIH.]

10:45

XX9. Effects of target phase in narrow-band frozen noise detection data. S. K. Isabelle and H. S. Colburn (Biomedical Engineering Department, Boston University, 110 Cummington Street, Boston, MA 02215 and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Hit and false alarm rates were estimated for detection of a 500-Hz tone target in each of ten samples of 1/3-oct noise centered at 500 Hz for both NoS π and NoSo conditions. The effects of target phase (0° and 90°) on hit rate were investigated for three subjects. Results for this narrow-band case were similar to previously reported results for wideband maskers [Gilkey *et al.*, *J. Acoust. Soc. Am.* 78, 1207–1219 (1985)]. Two differences were noted. First, for NoS π conditions, the effect of signal phase was not significant for the narrow-band maskers, whereas a small effect was reported for wideband maskers. Second, no significant correlation was observed between NoSo and NoS π results across masker waveforms for hit and

XX7. Psychophysical factors of spatial impression. Frans A. Bilsen, Herman W. Krussse, and Johan Raatgever (Applied Physics Department, Delft University of Technology, Delft, The Netherlands)

Spatial impression (or spaciousness) is an important perceptual attribute in room acoustics, implying (1) the sensation of a listener in a concert hall of being enveloped by the sound field and (2) the perceived broadening of the sound source. A psychophysical experiment will be reported using dichotic presentation of (filtered) white noise, in which the influence on spatial impression was investigated of three signal variables, viz., the interaural correlation, the center frequency, and the relative bandwidth of the filtered versions of the noise. Eight untrained subjects participated in a scaling experiment: They were presented 4 times with 80 (4•5•4) experimental conditions; subjective criterion was the perceived width of the sound image. The raw ordinal data were analyzed by canonic correlation analysis (CANALS); on the resulting data, ANOVA was applied. The main outcome of the experiment is that the interaural correlation and the center frequency are independent and adequate predictors of spatial impression. This result and the results of other experiments will be interpreted in the context of existing binaural theories. [Work supported by the Netherlands Organization for the Advancement of Pure Research (ZWO).]

false alarm rates, whereas significant correlations for both rates were reported for the wideband experiment. The lack of effect of target phase on hit rate in the narrow-band NoS π condition is inconsistent with models based on lateral position but consistent with the EC model and with models using the sum of the squares of interaural time and intensity differences. [Work supported by NIH.]

11:00

XX10. Lateralization on the basis of interaural envelope delays: The effect of additional frequency components. Raymond H. Dye, Jr. and Andrew J. Niemic (Parham Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, IL 60626)

A lateralization experiment was performed in which threshold interaural envelope delays were measured as a function of modulation frequency ($f_m = 25, 50, 100, 200, 300, 400,$ and 500 Hz) for three- and five-component complexes whose center frequency (f_c) was either 2000 or 4000 Hz. A two-alternative forced-choice procedure was used in which the envelope lagged to the right ear during one interval and to the left interval during the other. The level of each component was 50 dB SPL, and the signal duration was 200 ms with 10-ms linear rise-decay times. Thresholds obtained with three- and five-component complexes were quite similar regardless of f_m , a finding that seemed consistent with the notion that the envelope is extracted from components interacting within an auditory filter, with more distal components having no effect. Surprisingly, making the outermost sidebands ($f_c - 2f_m$ and $f_c + 2f_m$) diotic was found to severely impair one's ability to utilize envelope delays even when the f_m 's were quite high. These findings place in doubt the contention that the binaural auditory system extracts envelopes by monitoring the outputs of narrow-band auditory channels. [Work supported by NINCDS and AFOSR.]

11:15

XX11. Interaural envelope correlation and the high-frequency MLD. Leslie R. Bernstein (Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL 32611)

Data from a number of investigations suggest that listeners are extremely sensitive to interaural temporal disparities in the envelopes of

high-frequency, complex waveforms. In the present study, an attempt was made to assess whether this sensitivity could be the basis for MLDs obtained at high frequencies. An adaptive, two-alternative forced-choice task was used to estimate detection thresholds for a number of interaural configurations including NoSo, NoSm, and NoS π at a center frequency of 3571 Hz. In addition, a sinusoidally amplitude-modulated 3571-Hz tone whose carrier was interaurally in phase but whose modulator was interaurally phase-reversed was employed as a signal. The bandwidth of the

diotic masking noise ranged from 50 to 400 Hz. For all conditions, dichotic thresholds were accounted for quite well by assuming that, for each listener, detection occurs when the signal decreases the interaural correlation of the envelope of the noise to some threshold value. This finding as well as alternative explanations will be discussed in terms of modern models and theories of binaural hearing. [Research supported by Air Force Office of Scientific Research.]

FRIDAY MORNING, 20 NOVEMBER 1987

UMS 4 AND 5, 8:30 A.M. TO 12:00 NOON

Session YY. Underwater Acoustics IX: Bottom Interacting Ocean Acoustics I

David R. Palmer, Chairman

NOAA—Atlantic Oceanographic and Meteorological Laboratory, 4301 Rickenbacker Causeway, Miami, Florida 33149

Chairman's Introduction—8:30

Invited Papers

8:35

YY1. Acoustic issues from a marine surveyor's viewpoint. Wesley V. Hull (Charting and Geodetic Services, National Ocean Service, NOAA, Rockville, MD 20852)

The ships and instrumentation used in the National Ocean Service hydrographic and bathymetric surveying represent capital investment of several hundred million dollars and recurring annual costs of tens of millions. The efficiency in survey coverage, while maintaining required survey accuracy, is of utmost concern. Deficiencies in any of the several types of acoustic systems used in the surveys can and do contribute significantly to the loss of survey production. This paper describes briefly three types of survey missions, highlighting the acoustic systems used and the known or perceived acoustic phenomenological factors that contribute to lost production. These systems include dual frequency echosounders, multibeam swath sonar systems, and conventional side scan. The paper also discusses emerging technology, now under consideration, including both interferometric and multibeam side scan systems, again with emphasis on the critical acoustic phenomena. Finally, a new concept dubbed "the acoustic knife," for which an experimental project is now contemplated, is discussed.

9:00

YY2. Scattering of sound from the rough seafloor. T. K. Stanton (Department of Geology and Geophysics, University of Wisconsin, Madison, WI 53706)

The scattering of sound from the rough seafloor and resultant echo fluctuations are functions of seafloor geomorphology and sonar parameters such as beamwidth, frequency, and angle of incidence. The scattering is very difficult to describe analytically and quite often requires numerical and empirical techniques. Like most rough interfaces, the seafloor can be described by two classes of surfaces: (1) stochastic and (2) deterministic. For both types of surfaces, it is essential to model the roughness as being two-dimensional, i.e., not striated. Because of the randomness of the seafloor, the echo will fluctuate from ping to ping as the sonar moves over the floor. Patches or distinct geomorphological features will cause deviations in the statistical properties of the echoes. [Work supported by the ONR.]

9:25

YY3. Inverse methods in ocean bottom acoustics. George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A review is presented of exact and perturbative inverse methods used to determine the acoustic properties of the bottom. The discussion concentrates primarily on inversions for properties in the top few hundred meters of sediment using input data in the frequency range 25–500 Hz. The one-dimensional, horizontally stratified case is emphasized, although implications for problems of higher dimensionality are discussed. Descriptions of theoretical methods and experimental techniques as well as examples of inversions of synthetic and experimental data are presented.

YY4. Bottom interaction effects on broadband signals. Paul J. Vidmar^{a)} (Science Applications International Corporation, McLean, VA 22102)

This paper examines recent progress toward understanding the effects of the seafloor on low-frequency broadband signals. First, the major bottom interaction processes affecting low-frequency signals are reviewed and a simplified (single layer fluid) geoacoustic profile is shown to accurately predict their effects. A ray-based approach for simulating single bounce broadband signals is then discussed. The simulator uses the simplified geoacoustic profile to include the seafloor as part of the propagation medium. Under conditions for which the simplified profile accurately represents the acoustics of the seafloor, calculated waveforms agree well with acoustic data. Next, an approach for simulating multibounce signals interacting with a layered seafloor is described and used to show that layering in the seafloor is the likely cause of some characteristics of acoustic data that are not predicted by the single layer profile. Finally, the time spreading of broadband signals in an area with about 200 m of sediment cover is examined. Acoustics data have both small-scale (< 200 ms) and large-scale (> 1 s) time spreads. Viewing the data through several frequency bands reveals that small-scale time spreads are concentrated at high frequencies (consistent with reflection for near surface layering) while large-scale time spreads are composed primarily of low frequencies (consistent with interaction with the substrate). A simple ray analysis shows that the scattering from the region near the point of specular reflection cannot be the sole cause of the large-scale time spreading.^{a)} On leave from Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712.

Contributed Papers

10:15

YY5. Stress-induced anisotropy in sediment acoustics. R. D. Stoll (Lamont-Doherty Geological Observatory, Palisades, NY 10964)

A model to describe seismic wave propagation in both dry and water-saturated particulate materials is derived on the basis of the response of a regular array of like elastic spheres. The theories of Hertz and Mindlin are used to model the normal and tangential compliances at the intergranular contacts, and the overall response of the array is found to be governed by a set of nonlinear differential equations. These equations are integrated over a particular stress history that simulates the normal buildup and relief of geostatic stress (overburden pressure) in nature. Wave velocities are evaluated at various stages during this stress history and the response is found to be anisotropic with respect to both P and S wave velocities. In order to simulate the response of marine sediments, the effects of water saturation are investigated by using the compliances calculated for the dry skeletal frame in conjunction with the Biot theory. Finally, the predictions of the model are compared with recent laboratory experiments and found to be in general agreement. [Work supported by ONR, Code 1125 OA.]

10:30

YY6. Shallow seismoacoustic experiments using shear waves. R. D. Stoll, G. M. Bryan, R. Flood, D. Chayes, and P. Manley (Lamont-Doherty Geological Observatory, Palisades, NY 10964)

During the summer of 1986 a series of seismoacoustic experiments were carried out in shallow water off the New Jersey shore. The purpose of these experiments was to measure the geoacoustic properties of the ocean sediments that comprise the upper few hundred meters of the sediment column. Seismic sources and receivers were deployed at or very near the bottom in order to excite shear waves in the sediment and minimize the effects of interference from water-borne propagation. The experiments were performed at several sites where prior field work had established physical properties and a detailed profile of the sediments. By using conventional air guns deployed in an unconventional way, strong interface and diving shear waves were generated, and these data were inverted to obtain shear wave velocity as a function of depth. The inversion results were then compared with the predictions of a geoacoustic model that accounts for the effects of voids ratio, overburden pressure, and other physical parameters. The *in situ* measurements from experiments and the gradients predicted by the model were in good agreement, suggesting a strong dependence of velocity on overburden pressure near the water-sediment interface. [Work supported by ONR, Code 1125 OA.]

10:45

YY7. Quantitative experimental verification of seabed shear modulus profile inversions using surface gravity (water) wave-induced seabed motions. Mark Trevorrow, Tokuo Yamamoto, Mohsen Badiey, Altan Turgut, and Craig Conner (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, 4600 Rickenbacker Causeway, Miami, FL 33149-1098)

A previously developed theory enables seabed shear modulus profiles to be calculated through the combination of water wave-seabed interaction theories and geophysical inversion methods. A new instrumentation system, consisting of three orthogonally mounted accelerometers and a pressure sensor, has been developed to measure water wave-induced pressures and seabed motions. This new passive remote system provides a fast and convenient method to determine *in situ* sediment shear modulus with depth profiles, without disturbing the sediment or drilling. Experiments have been conducted in various sediments and water depths off the coast of New Jersey. Extensive experimental data reduction techniques are developed to deal with real ocean data. Actual experimental results are given, along with quantitative measures of uncertainty and resolution. Comparisons between the experimental shear modulus inversions, *in situ* sediment strength tests, and laboratory analysis of core samples show that this method can accurately predict shear modulus to depths of 50 m or more. [Work supported by ONR.]

11:00

YY8. Acoustic wave propagation through porous media with arbitrary pore-size distributions. Tokuo Yamamoto and Altan Turgut (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, 4600 Rickenbacker Causeway, Miami, FL 33149)

In the Biot theory, the effect of frequency on the oscillatory viscous forces within a porous medium is treated by replacing the kinematic viscosity ν by an oscillatory viscosity νF . Here, F is a function of angular frequency ω , the kinematic viscosity ν , and the single pore size a . In this paper, a mathematical expression of F is presented for arbitrary distribution of pore sizes that can be used in the Biot theory without modification. It is shown that porous media with a given permeability and porosity may be represented by an infinite number of pore-size distributions. The dispersion and attenuation of acoustic waves through such porous media are independent of the pore-size distribution at the low- and high-frequency limits, while they are strongly dependent on the pore-size distribution in the intermediate frequency range. For porous media with ϕ -normal pore size distributions having a given value of permeability, the maximum spe-

cific attenuation decreases and the bandwidth of dispersion increases with an increasing standard deviation of the pore-size distribution. Finally, the generalized Biot theory for arbitrary distribution of pore sizes is compared with the entire data of compressional wave attenuation through surficial marine sediments given by E. Hamilton. Comparison between his data and the predictions from this theory show good agreement. [Work supported by ONR.]

11:15

YY9. Effects of seabed properties on acoustic normal mode propagation in a stratified shallow ocean. Mohsen Badiéy (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, 4600 Rickenbacker Causeway, Miami, FL 33149)

The pressure field of a point source in a shallow stratified ocean over a nonhomogeneous anisotropic seabed is calculated using normal modes. The sediment is modeled using the Biot-Yamamoto theory of poro-elastic media. The influence of geoaoustic properties of sediments on low-frequency transmission loss is investigated. It is found that the variation of permeability has large influence on acoustic wave attenuation, whereas the elastic properties variation has lesser effect. Finally, the existing model is used in an attempt to extract the depth structure of permeability and porosity of the seabed. [Work supported by ONR.]

11:30

YY10. Synthetic modeling of shear waves in shallow marine sediments. J. A. Carter, G. H. Sutton, N. Barstow (Rondout Associates, Incorporated, P. O. Box 224, Stone Ridge, NY 12484), and J. I. Ewing (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

Full-waveform synthetic seismograms are being used to aid inversion for shear velocity and attenuation measured in shallow marine sediments off the coast of southern New Jersey in May 1986. Horizontally and vertically polarized shear waves were produced by a pair of air guns, fired alternately left and right, mounted on a sled. Initial studies demonstrated

that a simple shear source in a layered medium does not fully explain the observed data. Synthetic sources mimic the experimental source by combining a dipole shear, a torque, and an explosion source (in the water). This complex source is applied to different velocity/attenuation models, and results are compared to the data. The frequency band of the synthetics is 2–15 Hz, matching the predominant frequencies of the data; the phase velocity range is 0.05 to 1.0 km/s; all possible modes within these frequency and phase velocity ranges are computed. Varying thickness and velocity of a low-velocity layer both at the top and within the sediment section produce striking changes in the character of the seismograms. Normal or inverse dispersion characterize the late-arriving guided-wave portion of the synthetics. The early-arriving diving waves generally do not exhibit appreciable dispersion. [Research supported by ONR.]

11:45

YY11. Dynamic behavior of sediment and water-sediment interface. Homogenized process application to porous saturated media. C. Avallet (Metravib R.D.S., 64, Chemin des Mouilles, 69130 Ecully, France)

An application of the homogenized process to the dynamic behavior of marine sediments is presented here. The medium is considered as a porous viscoelastic solid saturated by a viscous fluid. Under the assumption of periodicity for pore structure (microscopic scale), the homogenization technique leads to the macroscopic description. The set of equations so obtained is similar to the classical Biot's results. The homogenized model is somewhat easier to use: A generalized Darcy's law governs the motion of the pore fluid relative to the solid [Auriault *et al.*, *J. Acoust. Soc. Am.* 77, 1641–1650 (1985)]. A reasonable assumption allows the computation of the frequency dependence of the permeability from its quasistatic value. Using the homogenized set of macroscopic equations, compressional and shear waves in porous media, surface waves at fluid-porous solid interface, and reflection properties of plane acoustic waves were calculated. Results are compared to those obtained by Biot's theory and to experiments. The aim of this work was to show the applicability of the model to unconsolidated sediments (high porosity silt and clay or sandy media). [Work supported by D.R.E.T.]

FRIDAY MORNING, 20 NOVEMBER 1987

TUTTLE SOUTH ROOM, 9:00 TO 11:40 A.M.

Session ZZ. Noise VI: Measurement Techniques and a Review of Noise Effects on Man

Julia D. Royster, Chairman

Environmental Noise Consultants, Inc., P. O. Box 144, Cary, North Carolina 27511

Chairman's Introduction—9:00

Contributed Papers

9:05

ZZ1. The influence of the center frequency of a narrow-band short-duration impulse noise on the growth of TTS. Chantal Laroche and Raymond Héту (Groupe d'acoustique de l'Université de Montréal, C.P. 6128, Succ. A, Montréal, Québec H3C 3J7, Canada)

The following study has contributed to the knowledge of the effect on human hearing of the spectral content of impulse noise, a parameter that has been little studied to date. In this study, signals were computer generated to obtain 1 octave band wide impulses from a $(\sin x)/x$ numerical

function. The center frequency was varied so as to generate four signals centered on 1, 2, 3, and 4 kHz. In this study, ITS growth curves were obtained with a target asymptotic effect of 8–12 dB after 30–45 min of exposure at a rate of 1 pps. The required level of exposure was determined to obtain this target effect. To reach the same level of ATS, the L_p of the 1-kHz signal had to be at least 5–7 dB higher than that of the 2-kHz signal. For two of four subjects, comparable levels were required for the 3- and 4-kHz signals to obtain the predetermined effect, but these were at least 5 dB below the corresponding level for the 1-kHz signal. These results demonstrate that the spectral content of impulse is a crucial factor in the prediction of the noxiousness of this type of noise. [Work supported by IRSST.]

9:20

ZZ2. Digital processing of aerodynamic signals. W. C. Meecham, E. Wildauer (Department of Mechanical, Aerospace, and Nuclear Engineering, University of California, Los Angeles, CA 90024), and S. A. McNerny (Aerospace Corporation, El Segundo, CA 90009)

Experimental data were obtained from tests performed in the NASA-Ames 7- \times 10-ft wind tunnel. A digitizing process for flow over a blunt-edged air foil was used to compare data with an analog system. The near-field surface-pressure signal on the wing and the farfield aerodynamic signal were cross-correlated. Digital filtering was applied to the data to suppress unwanted signals. The results of near-/farfield cross correlations show, using the digitized treatment, that there is substantial ringing, a relatively narrow-band source. Analog prefiltering exhibits a shift to lower frequency, due to frequency roll-off of the RC analog filters. It is shown that the clip-correlation (needed for analog) and the digitized full correlation methods give similar results. But the absolute value of the correlation is lost in clipping. [Work supported by NASA-Ames.]

9:35

ZZ3. Measuring the sound power of a noise source and detecting pure tones with a single instrument combining true 1/3 octave band digital filters and an FFT analyzer. Ole-Herman Bjor and Gustav Ese (Norwegian Electronics-Scantek, Inc., 51 Monroe Street, Suite 1603, Rockville, MD 20850)

In the past, there has been a requirement for determination of both the sound power of sources and an investigation of the pure-tone content. This is especially true in the computer and business equipment industry. Measurement of the sound power normally calls for standard (ANSI) class III 1/3 octave band filters, while pure-tone determination requires narrow-band (FFT) measurements; hence, two separate instrument systems were required. This paper describes a new instrument for these determinations in which the true standardized 1/3 oct filters and the FFT are

10:25

ZZ6. Evaluation of a hearing conservation program at a large industrial company. William W. Clark, Carl D. Bohl, Lisa S. Davidson, and Keith A. Melda (Central Institute for the Deaf, 818 S. Euclid, Saint Louis, MO 63110)

The hearing conservation program (HCP) of a large industrial company was evaluated by comparing the audiometric data of two groups of workers from three geographically separate plants. Group 1 was selected from individuals employed in areas where the time-weighted average (TWA) sound level exceeded 90 dBA and group 2 from individuals with TWAs of less than 85 dBA. Differences between mean STS values for groups and plants were the main criteria used in making judgments on the HCP effectiveness. In addition, age, sex, years of service, and nonoccupational noise exposure were evaluated. A significant difference in mean STS was found between the two groups; however, the "nonoccupationally exposed" group had more STS than expected. The male-female distribution, previous hearing loss, and nonoccupational exposure were judged to be contributing factors in the differences that existed between plants and groups. [Work supported by Grant OH 02128 from NIOSH.]

10:40

ZZ7. Sound propagation and community noise exposure considerations for enroute noise of advanced turboprop aircraft. John Wesler (Wyle Laboratories, 2001 Jefferson Davis Highway, Suite 701, Arlington, VA 22202) and Louis C. Sutherland (Wyle Laboratories, 128 Maryland Street, El Segundo, CA 90245)

combined. The procedure for automatically identifying prominent discrete tones per ANSI S12.10 is discussed.

9:50

ZZ4. Noise source identification on an IBM typewriter using sound intensity technique. Li J. Zeng and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

Noise source identification was carried out on an IBM ball element electric typewriter. The selective operation method was used to survey the vibration and sound pressure of different typewriter sources. The sound power radiation of different noise sources was measured using the two-microphone sound intensity technique. The measurements showed that the impulse radiation of the ball element striking the platen is the highest noise source, and the difference between the results with and without paper is small. Several indexes that suppose to qualify the accuracy of sound power measurements using the sound intensity technique will be discussed. [Work supported by IBM, Austin, TX.]

10:05

ZZ5. Determination of loudness with a programmable real-time analyzer. M. Brechbuehl and Svein Arne Nordby (Norwegian Electronics-Scantek, Inc., 51 Monroe Street, Suite 1603, Rockville, MD 20850)

An efficient program for determination of loudness as specified by Zwicker [E. Paulus and E. Zwicker, "Computer programs for calculating loudness from 1/3 octave band levels or from critical band levels" (in German), *Acustica* 27, 253 (1972)] is made to operate on a portable real-time analyzer. The program controls the measurement and calculates the loudness as per ISO R 532B. The user may view the results graphically on the RTA's black and white monitor or in full color on an external monitor. This completely automatic system operates without the use of an external computer due to the unique internal BASIC programmability of the real-time analyzer used. The user may easily modify the program to meet his/her own requirements incorporating EPNdB, Stevens Loudness, etc.

10:20-10:25
Break

The sensitivity of predicted ground noise levels for advanced turboprop aircraft to changes in atmospheric absorption losses is reviewed, and the resulting community noise environments evaluated in terms of possible criteria for acceptable levels. Atmospheric absorption coefficients at high altitudes are subject to possible errors of over $\pm 70\%$ due to the current uncertainty in atmospheric absorption prediction models at the very low humidities involved. However, the integrated absorption loss over the entire propagation path is nearly independent of the particular prediction model chosen. The humidity and temperature structure of the atmosphere is a more significant source of variation in the total atmospheric absorption loss. Based on a reference profile of humidity content versus altitude, and existing models for a standard atmosphere, estimates of enroute noise of advanced turboprop aircraft are made and compared with available data. The noise exposure on the ground is considered in terms of both the maximum sound level and sound exposure level for one overflight, and the day-night average sound level for various scenarios of daily operations. These estimates of the resulting community noise exposure are compared to possible criteria for acceptable levels for single and multiple aircraft flyovers. [This work was supported in part by the Federal Aviation Administration.]

10:55

ZZ8. A theoretical interpretation of a dosage-effect relationship for the prevalence of annoyance in a community. Sanford Fidell, David M. Green, and Theodore Schultz (BBN Laboratories Incorporated, P. O. Box 633, Canoga Park, CA 91304-0633)

A dosage-effect relationship by Schultz ["Synthesis of social surveys on noise annoyance," *J. Acoust. Soc. Am.* 64, 377-405 (1978)], that

relates community noise exposure to the prevalence of annoyance, has become a widely used tool in assessing environmental noise effects. Because Schultz's relationship is applied to a broad range of exposure conditions, and because it is entirely empirical, its application to specific situations is frequently challenged. A simple mathematical model can account not only for the shape of the relationship synthesized by Schultz, but also for the differences between Schultz's relationship and that proposed by Kryter ["Community annoyance from aircraft and ground vehicle noise," *J. Acoust. Soc. Am.* 72, 1222-1242 (1982)]. A single parameter, representing response bias, suffices for this purpose. [This work was sponsored by the U.S. Air Force Noise and Sonic Boom Impact Technology program under Contract F33615-86-C-0530.]

11:10

ZZ9. Effects of hearing protectors on head diffraction. John R. Franks (Bioacoustics and Occupational Vibration Section, Physical Agents Effects Branch, Division of Biomedical and Behavioral Science, National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, OH 45226)

The diffraction by the head and pinna of sound incident upon the ear has been well documented. However, the modification of head diffraction patterns by the placement of hearing protectors in, or over, the ears has not been investigated. Polar sensitivity plots were obtained in an anechoic sound field from the right ear of the Knowles Electronics Mannequin for Acoustic Research (KEMAR). Test frequencies ranged from 100-

10 000 Hz in steps of 1/3 octave. Plots were obtained in the horizontal plane for three conditions: (1) ear open; (2) ear occluded with a malleable foam-type ear plug; and (3) ear covered by a conventional ear muff. Polar data were converted to field-to-drum transfer functions for every 45° change in angle of signal incidence for each condition. The diffuse-field ear-open transfer functions will be compared to previously published functions for KEMAR. The effects of ear occlusion with muffs, caps, and plugs as seen by attenuation and canal resonance shift will be discussed.

11:25

ZZ10. A theoretical model of the annoyance of individual noise intrusions. Sanford Fidell, David M. Green, and Karl S. Pearsons (BBN Laboratories, Inc., P. O. Box 633, Canoga Park, CA 91303)

Self-reports of noise-induced annoyance can be modeled as the product of a rational, decisionlike process in which people discriminate among annoying and nonannoying noise intrusions heard in the presence of a reference distribution of noise exposure. This view, developed under the sponsorship of the U.S. Air Force Noise and Sonic Boom Impact Technology program, can assign explicit roles to both acoustic and nonacoustic variables in a probabilistic decision-making process. The model, expressible in spreadsheet form, permits identification of two components of annoyance judgments: direct annoyance, based on sensitivity to the immediate characteristics of acoustic signals, and response bias. The model also provides insight into the extent to which acoustic variables may be considered to cause annoyance.

FRIDAY MORNING, 20 NOVEMBER 1987

UM AUDITORIUM, 9:00 A.M. TO 12:00 NOON

Session AAA. Speech Communication VII: Cross Language Studies of Production

George D. Allen, Chairman

Department of Audiology and Speech Sciences, Purdue University, West Lafayette, Indiana 47907

Contributed Papers

9:00

AAA1. The coordination of glottal with oral articulations in Icelandic and Hindi. John Kingston (DMLL, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

In most consonants produced with glottal abduction, the opening is largely or entirely contained within the accompanying oral articulation. Furthermore, the peak glottal opening occurs at a constant interval from the beginning of the oral articulation in unaspirated stops and voiceless fricatives and at a constant interval from the release in aspirated stops, indicating tight coordination with the oral articulation [A. Löfqvist and H. Yoshioka, "Interarticulator programming in obstruent production," *Phonetica* 38, 21-34 (1981)]. Icelandic and Hindi possess stops in which the glottal opening takes place largely outside the oral closure; in Icelandic preaspirated stops, abduction precedes the closure and in Hindi breathy voiced stops, abduction follows the closure. Both languages also have (post) aspirated stops with which the stops of interest can be compared. This study examined whether external glottal articulations are also tightly coordinated with the oral articulation of the consonant. One female speaker of each language has been recorded so far, producing words containing the stops of interest at different rates and with stress either on the syllable containing the stop or not. If the oral and glottal articulations are coordinated with one another, then changes in their individual durations due to changes in rate or stress location should covary. The duration of both pre- and post-aspiration in Icelandic covaried with the duration of the adjacent vowel, and the duration of preaspiration also covaried with the following stop closure, but the duration of postaspiration did not co-

vary with the preceding stop closure. In Hindi, the duration of both breathy voice and (post) aspiration covaried with the duration of the preceding stop closure but the duration of neither covaried with that of the following vowel. Both kinds of Hindi stops are like those observed in other languages, but the relative timing of glottal and oral articulations in the Icelandic stops is clearly different. In that language, the oral articulation with which the glottal articulation is coordinated is the vowel or the vowel and consonant combined.

9:15

AAA2. Temporal effects of geminate consonants and consonant clusters. Margaret H. Dunn (Yale University and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Current phonological theory analyzes geminate consonants as sequences of adjacent timing slots that completely share features while closing one syllable and opening the next. This analysis predicts that the temporal organization of utterances with geminate consonants is parallel to that of utterances including heterosyllabic consonant clusters. This prediction was tested using Finnish and Italian, two languages with very different temporal organizations. Five Italian and four Finnish subjects read lists of nonsense words including bilabial geminates and both homorganic and heterorganic clusters. The hypothesized equivalence of geminates and clusters was tested by examining the amount of closed syllable vowel shortening exhibited in the two environments. The results did show that geminates patterned with clusters, with respect to preceding vowel

duration in both languages, but closed syllable vowel shortening was exhibited only in the Italian. This result suggests that the syllable may be used as a unit of temporal organization in Italian, but some other units of temporal organization must be proposed for Finnish. [Work supported by NIH Grant NS-13617.]

9:30

AAA3. VOT variability in stop-consonant productions by bilingual speakers of English and Mandarin Chinese. Judith L. Lauter and Fang Ling Lu (Department of Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721)

It has been observed [e.g., L. Williams, *Percept. Psychophys.* 21, 289–297 (1977)] that some bilingual speakers of Spanish and English will use language-appropriate VOT values in elicited sets of English and Spanish stops. As part of an ongoing cross-language project in repeated-measures acoustic phonetics, six native speakers of Mandarin Chinese (three females and three males) were asked to produce repetitions of English and Chinese stops in a nonsense/ta-'Ca/context. For the Chinese stops, five separate series were collected, with the stressed syllable pronounced on each of five Chinese tones. Productions were digitized via pulse code modulator, stored on video tape, and VOT segments identified and measured using waveform analysis software developed in our laboratory. Measurements of *mean* VOTs for each talker indicate that English and Mandarin Chinese are similar in range of absolute VOT values, clear VOT distinctions between voiced/voiceless stops, and inconsistent use of VOT to distinguish place of articulation. Further, VOT *variability* patterns suggest that for some of these talkers, control of VOT timing may be sensitive to tonal context, and this sensitivity may be reflected in productions of English as well as Chinese stops. [Work supported by AFOSR.]

9:45

AAA4. Stop consonant voicing in Malay-English bilinguals. Grace H. Yeni-Komshian and Inderjit K. Bhathal (Department of Hearing and Speech Sciences, University of Maryland, College Park, MD 20742)

The productions of two groups of Malay-English bilinguals were investigated. The subjects, native speakers of Malay, were asked to read Malay and English words that differed in voicing of initial and final stop consonants. The four subjects in the Malaysia group resided in Malaysia and were not exposed to a predominantly English speaking environment, and the two subjects in the U. S. group resided in the U. S. for 5 years. Results on word initial stops are reported here. In Malay, voiced initial stops are characterized by voicing lead and voiceless stops by short voicing lag. Comparison of the two bilingual groups revealed that, although there were no significant differences in the voiced productions, there was a shift towards a more Englishlike production for the voiceless stops by the U. S. group. Analysis of the English words showed no differences in the production of the voiced stops; however, the U. S. group produced the voiceless stops with significantly longer VOT values than the Malaysia group. None of the subjects showed a shift in voicing characteristics as a function of language. These results suggest that native Malay productions in bilinguals, residing in the U. S., were altered in a direction that is characteristic of English.

10:00

AAA5. Output constraints and cross-language differences in coarticulation. Sharon Y. Manuel (Bldg. 36-529, Massachusetts Institute of Technology, Cambridge, MA 02139)

Coarticulatory patterns have been shown to vary from language to language, from speaker to speaker, and from one style or rate of speaking to the next. Many of these differences might be explained by assuming (1) that phones are associated with ranges of acceptability, that is, "acoustic output constraints," which must not be violated by coarticulation-induced variability and (2) that these output constraints vary from language to language, speaker to speaker, etc. For this notion to have any *predictive* value, it is necessary, in turn, to be able to predict, to some extent, just what the acoustic output constraints are, and how they might vary. It might be expected, for example, that output constraints relax in

casual speaking styles. The paper is particularly concerned with output constraints that can be understood in terms of systems of phonemic contrast. Specifically, it is predicted that when the phonemes of a language are widely spread out in the acoustic space, then the acoustic output constraints on them are less stringent than if the phonemes were crowded together. Data supporting this view will be presented for vowel-to-vowel coarticulation in three Bantu languages. [Work supported by NIH to Haskins Laboratories and NIH to M. I. T.]

10:15

AAA6. Support for the frame model in Turkish and in English. Suzanne E. Boyce (Haskins Laboratories, New Haven, CT 06510)

The look-ahead and the frame model of anticipatory coarticulation make different predictions about lip movement in a word such as /ustu/. The look-ahead model predicts that rounding from the second vowel will spread into the intervocalic consonants, producing a single movement with a long sustained phase; the frame model predicts two distinct gestures for the rounded vowels whose tails may overlap somewhat, producing two peaks with a "trough" between. In earlier work, it was shown that for nonsense words, such as /kuktluk/, English speakers produce troughs, while Turkish speakers show plateau-like patterns. Does this mean that Turkish and English speakers employ, respectively, look-ahead and frame strategies for coarticulation? Alternatively, the frame model holds for Turkish, but Turkish rounding gestures are larger and longer, overlapping for a greater proportion of their tails. An additive model of overlap could then produce a plateau-like pattern. Speakers' movement traces from /kiktlik/ and /kuktluk/ were added and the result compared with those for /kuktluk/; a good fit for Turkish speakers was taken as evidence for the frame model. Further, the proportion of rounding during the consonant interval was different for the two languages. Both tests support the frame model. [Work supported by NIH.]

10:30

AAA7. Quantitative characterization of degree of coarticulation in CV tokens. Abigail C. Cohn (Department of Linguistics, University of California, Los Angeles, CA 90024-1543)

Languages are often assumed to show greater or lesser degrees of coarticulation [P. Ladefoged, in *Papers in Linguistics and Phonetics in Memory of Pierre Delattre*, edited by A. Waldman (Mouton, The Hague, 1972), pp. 273–286], yet no quantitative measure exists for characterizing degree of coarticulation. In this study, the development of a metric for degree of influence of a vowel on a preceding stop consonant has been undertaken. For each language in the study, LPC spectra were made at the burst onset of initial stops followed by a vowel. Templates were made of these spectra when the vowel was /a/. The stop spectra with other vowels were tested against these templates to determine a percent fit. Languages differ in their percent fit; a low percent fit indicates that vowels have a great effect on preceding consonants and a high percent fit indicates that they do not. Kana, a language of Nigeria which gives the auditory impression of having a high degree of coarticulation, is being compared to other languages including Russian, a language with secondary articulations. [Work supported by NSF.]

10:45

AAA8. Timing of contextual nasalization in two languages. Marie K. Huffman (Department of Linguistics, University of California, Los Angeles, CA 90024-1543)

Coarticulatory patterns in nasality in two West African languages—Akan and Efik—will be described. These languages make different phonological use of the feature (nasal). Nasality is determined from nasal and oral airflow as recorded with a Rothenburg split mask. The focus is on the timing of contextual nasalization; in particular, the coarticulatory effects of nasal consonants on adjacent vowels that are not contrastively (+ nasal). These effects are said to vary depending on the phonological use of nasality in a language. In Akan, there is an oral/nasal contrast on vowels,

even in the environment of nasal consonants; thus contextual nasalization may be limited, so as to preserve the phonemic contrast. In Efik, there is no such contrast on vowels, so contextual nasalization of vowels may be more extensive, beginning earlier, and ending later, as has been reported elsewhere for English. Data for several speakers of each language will be discussed. [Work supported by NSF.]

11:30

AAA9. Cross-linguistic articulatory models of vowels. Michel T. T. Jackson (Phonetics Laboratory, Department of Linguistics, University of California, Los Angeles, CA 90024-1543)

The mathematical and phonetic foundations of a family of techniques for quantitatively analyzing articulator positions during speech production will be discussed in terms of a small number of modes that span the observed space of articulatory variation. Particular articulatory positions are represented as weighted combinations of the modes; various naturalness conditions that depend on having multispeaker data can be used to resolve mathematical indeterminacies in the choice of modes. Results using measurement of articulator positions from sagittal x rays of English, Icelandic, Spanish, and Akan vowels suggest that the basis needed for any one language is of relatively small dimensionality. Typically, only two or three modes suffice to generate the observed range of articulator positions. However, not all languages have the same modes of articulator positioning. Thus a cross-language modal analysis requires higher dimensionality, i.e., more modes. In conclusion, there are systematic differences between languages in the functional organization of a set of articulatory primes. These differences in functional organization lead to the observed cross-language differences in modal organization. [Work supported by NSF.]

11:15

AAA10. Acoustic correlates of TENSE-LAX vowels and ATR vowels. Mona Lindau-Webb (Phonetics Laboratory, University of California, Los Angeles, CA 90024-1543)

Germanic vowel systems typically include two sets of vowels, a long, tense set and a short, lax set. Formant charts of English, German, Dutch, and Swedish show that lax high and mid vowels tend to have a more centralized position than their tense counterparts. Tense and lax low vowels do not show any such systematic relationships in the different languages. In African languages with vowel harmony, the harmonizing sets are sometimes labeled tense-lax, although a more appropriate label is advanced-retracted tongue root (ATR). Formant charts of Akan, Dholuo, and Agwagwune (a Nigerian language) were compared to the Germanic ones. The tense-lax and + / - ATR features behave very similarly for front vowels. Back - ATR vowels, however, tend to have a lower F_2 than their + ATR counterparts, making them more peripheral, not more central like back lax vowels. Thus tense and ATR are not the same acoustically. Although these two features do not occur contrastively in any language, they do possibly both occur in rules within a language. Agwagwune has centralized allophones in closed syllables. These are not central-

ized like the lax vowels in Germanic languages, but could still be described with a feature lax.

11:30

AAA11. Cross language study of the effects of voiced/voiceless consonants on the vowel voice source characteristics. Ailbhe Ní Chasaide and Christer Gobl (Centre for Language and Communication Studies, Trinity College, Dublin 2, Éire and Department of Speech Communication and Music Acoustics, KTH, Box 70014, S-100 44 Stockholm, Sweden)

Source characteristics of a vowel may differ according to the voiced/voiceless nature of adjacent consonants. The postvocalic consonant could be particularly crucial, as vocal fold abduction for a voiceless consonant may be initiated considerably before oral occlusion. Here, CV(:)C utterances (where C = voiced/voiceless labial stop or fricative) were analyzed for female and male speakers of Swedish, English, and French. Characteristics of the voice source were measured from inverse filtered data using the four-parameter LF model developed at the KTH. Airflow recordings with a Rothenberg mask allowed inferences regarding incomplete vocal fold closure during the vowel and at onset and offsets. Results indicate major effects of a following consonant; the later part of the vowel preceding a voiceless consonant shows a marked drop in excitation strength and a steeper spectral slope, as might be expected when the vocal folds are opening but vibrating. A spectral consequence of this abducting gesture is a widening of the F_1 bandwidth and an upward shift in formant frequencies. These effects were much less in evidence in French than in Swedish and English. The preceding consonant had comparatively little effect; full excitation strength is achieved almost immediately.

11:45

AAA12. An investigation of tag intonation in English. Richard W. Sproat and Shirley A. Steele (AT&T Bell Laboratories, 2D-443, 600 Mountain Avenue, Murray Hill, NJ 07974)

Tags frequently occur in English text. Common examples include "Are you coming?" he said," where "he said" is an attributive tag, and "I bought a new car, Manny," where "Manny" is a vocative tag. Because of the frequency of tags, it is important to have an adequate treatment of their intonation for more natural sounding text-to-speech synthesis. This paper examines the intonational properties of tags in American English and discusses their phonological characterization. Investigated instrumentally are the F_0 patterns employed by speakers producing tags in question and statement contexts ("Are you hungry?" he said" versus "I am hungry," he said"). A phonological analysis is proposed for such tags building on work by Pierrehumbert ["The phonology and phonetics of English intonation," Ph.D. dissertation, MIT (1980)] and Beckman and Pierrehumbert ["Intonational structure in Japanese and English," *Phonology Yearbook* 3, 255-309 (1986)]. Finally the incorporation of these results into the text-to-speech system under development at Bell Laboratories is discussed, and the resulting improvement in the quality of synthesis is demonstrated.

Session BBB. Psychological and Physiological Acoustics IX: Physiological Acoustics

Özcan Özdamar, Chairman

Department of Biomedical Engineering, University of Miami, Coral Gables, Florida 33124

Contributed Papers

1:30

BBB1. Phase-lock of spontaneous oto-acoustic emissions to a cubic difference tone. Pim van Dijk and Hero P. Wit (Institute of Audiology, P. O. Box 30.001, 9700 RB Groningen, The Netherlands)

If a single external tone is used to study phase-lock of a spontaneous oto-acoustic emission, it is difficult to separate this tone from the emission signal. Therefore, two tones with frequencies f_1 and f_2 and levels L_1 and L_2 were presented to ears having a spontaneous oto-acoustic emission. Here, f_1 and f_2 produced a $2f_1 - f_2$ distortion product close to (< 50 Hz) the emission frequency f_0 . Frequency spectra and zero-crossing time trend recordings were obtained for various values of L_1 , L_2 , f_1 , and f_2 . When the L_1 and L_2 were sufficiently high, the emission was phase locked to the $2f_1 - f_2$ tone generated in the ear. When L_1 and L_2 were small, the emission was sustained at f_0 . At intermediate levels, the emission is not constantly locked to $2f_1 - f_2$, but occasionally jumps to f_0 . The behavior described above is identical to that of a self-sustained oscillator in the presence of noise, driven by an external tone [R. L. Stratonovich, *Topics in the Theory of Random Noise* (Gordon and Breach, New York, 1963), Vol. II, Chap. 9]. [Work supported by Z. W. O.]

1:45

BBB2. Effects of frequency separation of primary tones on the amplitude of acoustic distortion products. Frances P. Harris (Department of Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721), Barden B. Stagner, Glen K. Martin, and Brenda L. Lonsbury-Martin (Department of Otorhinolaryngology and Communicative Sciences, Baylor College of Medicine, Houston, TX 77030)

The effects of frequency separation of equilevel primary tones, f_1 and f_2 , on the amplitude of distortion-product emissions (DPEs) at $2f_1 - f_2$ was investigated in 10 ears of five humans with normal hearing. The DPEs at 1, 2.5, and 4 kHz in response to primaries at 65, 75, and 85 dB SPL were generated with f_2/f_1 ratios varying in 0.02 steps from 1.01 to either 1.41 (4 kHz), 1.59 (2.5 kHz), or 1.79 (1 kHz). Results indicated that, in generating the maximum DPE amplitude, $f_2 - f_1$ frequency separation was systematically related to stimulus frequency and level. Thus, for all levels of stimulation, f_2/f_1 ratio was inversely related to DPE frequency so that higher ratios were most effective in generating DPEs at 1 kHz, while lower ratios elicited large emissions at 2.5 and 4 kHz. In addition, for all three DPEs, in obtaining maximum responses, primary-tone level was directly related to f_2/f_1 ratio in that lower ratios were most efficacious in generating DPEs to 65-dB stimuli, while higher ratios elicited larger DPEs in response to 75- and 85-dB primaries. On the average, however, across all frequency-amplitude stimulus conditions, maximal DPE amplitude was best obtained using an f_2/f_1 ratio of 1.23. Overall, DPE findings describing the outcome of varying level, frequency separation, and frequency region of primary tones were consistent with those reported in the psychophysical literature for normal-hearing humans. [Work supported by NINCDS.]

2:00

BBB3. Spontaneous oto-acoustic emission from a chinchilla ear following exposure to noise. William W. Clark and Martha Solomonson (Central Institute for the Deaf, 818 S. Euclid, Saint Louis, MO 63110)

A spontaneous oto-acoustic emission (SOAE) was observed in one chinchilla after two exposures to an octave band of noise (OBN) centered

at 0.5 kHz, 95 dB SPL, each for 9 days. This SOAE, not present before the second exposure, was a narrow-band signal at 2.2 kHz that varied from 10–26 dB SPL across recording sessions. The SOAE has remained stable over the 3-year period since it was discovered. During the past year, this animal has been trained behaviorally so that detection and discrimination thresholds can be determined in the region near the SOAE. Measures of auditory sensitivity at quarter-octave intervals between 0.125 and 16.0 kHz revealed a hearing loss (*re*: normal chinchilla) of 22 dB centered at 2.37 kHz. However, thresholds determined at 10-Hz intervals between 1600 Hz and 2800 Hz showed that the SOAE was correlated generally with a region of enhanced sensitivity in the microstructure of the audiogram. However, in the immediate vicinity of the SOAE thresholds were elevated by 5–10 dB; addition of a tonal masker (2.2 kHz at 5–20 dB SPL) had no effect on the shape of the audiogram. [Work supported by Grant OH 02128 from NIOSH.]

2:15

BBB4. Interaural phase and level effects on binaural interaction component (BIC) of the brain-stem auditory evoked potential. Ballachanda B. Bopanna and George Moushegian (Callier Center UTD, 1966 Inwood Road, Dallas, TX 75235)

Electrophysiological studies have elegantly demonstrated that neurons in lower brain-stem nuclei are exquisitely sensitive to the parameters of sound that mediate localization and lateralization phenomena. In the present study, the binaural interaction component (BIC), derived from brain-stem auditory evoked potentials, was obtained using low-frequency stimuli with various combinations of time and level differences delivered through earphones. The aim has been to determine whether these volume-conducted multineuronal responses reveal any of the neural mechanisms of lateralization. The findings show that characteristics of the binaural interaction component, e.g., latency, amplitude, and waveform morphology, evoked by low-frequency tone bursts, are differentially altered with changes in interaural phase. The effects on the binaural interaction component with interaural level differences were also substantial but not similar to the changes produced with phase. From this research, it has become apparent that, given appropriate recording and stimulating parameters, many nuances of binaural hearing may be neurophysiologically addressed in studies of volume-conducted evoked and derived potentials.

2:30

BBB5. AP and ABR tuning curves: A comparative study. Abdelhamid A. Elshintinawy and Paul J. Abbas (The University of Iowa, Department of Speech Pathology and Audiology, Wendell Johnson Speech and Hearing Center, Iowa City, IA 52242)

Whole-nerve action potential (AP) and auditory brain-stem response (ABR) were recorded from anesthetized cats. Response amplitude and latency to a probe stimulus were measured and tuning curves were constructed using three masking paradigms; forward, simultaneous (a tone burst masker overlapping the probe), and continuous (a continuous masker tone). AP and ABR tuning curves were generated on the basis of criterion changes in both amplitude and latency of the response. Within each of the masking techniques, tuning curves developed from the amplitude and latency criterion changes of the AP and the ABR showed no significant differences. The only significant differences were measured across masking conditions. Forward masking tuning curves were found to have steeper high-frequency slopes, higher Q_{10} values, and lower sensitiv-

ity indices (defined as the difference between probe level and the level at the tip of the tuning curve). Low-frequency slopes and the tip-to-tail ratios showed no significant differences. [Work supported partially by the Deafness Research Foundation.]

2:45

BBB6. Fiber tract model of auditory brain-stem response generation. Özcan Özdamar, Rafael E. Delgado (Department of Biomedical Engineering, University of Miami, Coral Gables, FL 33124), and Ferdinando Grandori (Polytechnic of Milan, Milan, Italy)

A fiber tract model for the generation of auditory brain-stem responses (ABRs) is developed using volume conducted potentials in a spherical conducting medium. This model proposes that the fast components of the ABR are generated by the synchronized propagation of the action potentials traveling along the auditory brain-stem pathways. The model represents an action potential with up to 100 current dipoles traveling along a pathway corresponding to a known auditory brain-stem tract. Coordinates of the major fiber tracts are obtained from human brain atlases and adapted to a spherical head model. Computations assuming classical interconnections between brain-stem nuclei show that such synchronously propagating action potentials can generate waveforms similar to recorded ABRs. According to this model, positive peaks correspond to the depolarizing wave front of the action potential while negative valleys are generated with the repolarizing phase.

3:00

BBB7. Steady-state evoked potentials in sleeping humans to continuous modulated tones. Lawrence T. Cohen and Field W. Rickards (Department of Otolaryngology, University of Melbourne, Parkville, Australia 3052)

At a previous meeting of the society, Rickards and Clark [J. Acoust. Soc. Am. Suppl. 1 72, S54(1982)] showed that steady-state evoked potentials could be recorded from alert humans in response to amplitude-modulated tones over a wide range of modulation frequencies. A subsequent paper has shown that, while there is an amplitude response maximum for alert subjects at a modulation rate of approximately 40 Hz and this frequency is thus appropriate for optimum threshold detection, different circumstances apply for sleeping subjects. Sleeping subjects differ from alert subjects in both amplitude response and background EEG noise, as functions of modulation frequency. A detection efficiency function reveals that during sleep the optimum modulation frequency, dependent on stimulus carrier frequency, lies well above 40 Hz. For lower carrier frequencies a peak occurs at about 90 Hz, while at higher frequencies the optimum modulation frequency is as high as 150 Hz. These higher modulation rates allow more efficient detection than does 40 Hz. Stimuli with high modulation rates have been used successfully with sleeping subjects including neonates.

3:15

BBB8. AGC behavior of a recovery model for auditory neuron firings. Timothy A. Wilson (Room 36-547, Department of Electrical Engineering and Computer Science, and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

In a recovery model for auditory neuron firings [Westerman and Smith, J. Acoust. Soc. Am. 81, 680-691 (1987); Johnson and Swami, J. Acoust. Soc. Am. 74, 493-501 (1983)], the firing history manifests itself as a reduction in the instantaneous probability of a neural event. In this paper, it is shown that the ensemble behavior of such a model may be interpreted as that of a feedback automatic gain control (AGC). The AGC input is the excitation function and its output is the ensemble-average firing rate. The impulse response of the gain-control filter is a superposition of time-invariant and time-varying components that are related to the recovery function's absolute- and relative-refractory intervals, respectively. In response to a steplike input, the model's output resembles the firing-rate response of an auditory neuron to a tone burst: Both contain an exponentially decaying rapid transient whose time constant decreases as the input size increases. [Work supported by DARPA under contract N00014-82-K-0727, monitored through the Office of Naval Research.]

3:30

BBB9. Evaluation of an eight-channel scala tympani electrode with single unit recording in the auditory nerve. David. H. Liang (AEL 128, Department of Electrical Engineering, Stanford University, Stanford, CA 94305)

In order for a multichannel cochlear prosthesis to be more effective than a single-channel prosthesis, the different channels of the stimulating electrode must be able to activate separate populations of the neurons in the auditory nerve. Earlier animal studies [C. van den Honert and P. Stypulkowski, Hear. Res. 14, 225-243 (1984); H. S. Lusted, Ph.D. thesis, Stanford University, 1980] have resulted in conflicting conclusions about whether this occurs for monopolar stimulation. To address this question, an eight-channel electrode array was acutely placed in the scala tympani of seven cats. Recordings were then made in the VIIIth nerve, of single unit responses to electrical stimulation from each of the channels in the array. For each unit isolated, threshold current for biphasic monopolar stimulation was determined for each of the electrode channels. Some units were found to be highly selective in their response to the different channels of the electrode array, with the most effective electrode having a threshold lower than an adjacent electrode by up to a factor of 5 times. Adjacent electrodes were either 1.0 or 1.5 mm apart. Other units were found not to be very selective in their response. This suggests that some degree of spatial selectivity is achievable with monopolar stimulation. [Work supported by NIH.]

3:45

BBB10. Electromagnetic basis of acoustic perception. I. Biological structures. Glenn M. Cohen (Department of Biological Sciences, Florida Institute of Technology, Melbourne, FL 32901)

Although hearing aids can compensate for some hearing impairments, they cannot effectively compensate for impaired functions resulting from major losses of hair cells and/or nerves. Cochlear implants are limited in their effectiveness by the number of functional cochlear neurons, surgical techniques, and current technology. Electromagnetic waves, a new approach, offer the potential of bypassing the severely impaired cochlea and directly stimulating the higher auditory centers, such as the auditory cortex [P. L. Stocklin and B. F. Stocklin, TIT J. Life Sci. 9, 29-51 (1979)]. This approach is based upon the fact that excitable tissues may emit electromagnetic waves during the depolarization cycle due to energy changes in the integral membrane proteins. In this paper, the auditory application of electromagnetic waves is introduced by examining the biological structures important to acoustic perception.

4:00

BBB11. Electromagnetic basis of acoustic perception. II. Physical mechanisms. John G. Clark (Mentec, Inc., 8940 S.W. 129 Terrace, Miami, FL 33176)

A study of physical parameters leads to the conclusion that the human skull, with brain tissue included, is a cavity that will support electromagnetic standing wave fields in the frequency range between 200 MHz and 3 GHz. This frequency range coincides with allowable transitions in the rotational energy states of integral proteins that are embedded in nerve membranes. This frequency overlap permits the integral proteins to couple energy between propagating action potentials and standing electromagnetic fields within the skull. This is a crucial physical mechanism in a full theory of brain functioning which accords standing EM fields a central role in acoustic perception. All physical mechanisms will be discussed and related to biological structures in the brain. A key prediction of the theory is that the frequency of the lowest EM standing wave mode for any individual can be predicted simply with knowledge of the cephalic index. Recent measurements made on seven individuals confirm the theory. [Work sponsored by Mentec, Inc.]

4:15

BBB12. Effect of acoustic tone perception on several higher modes of brain-generated electromagnetic waves: Preliminary evidence. Philip

Brain-generated microwaves were detected and measured for each of two subjects for several higher modes in the frequency range of the 7th–13th primary mode of the adult human brain (1200–1900 MHz). Differences in mode amplitude of 5–8 dB were measured with a loud acoustic

tone stimulus ON and OFF. Higher mode amplitude change occurs immediately upon tone stimulus start or stop. In order to increase higher mode power well above ambient noise, a zero mode “pumping” procedure was used. Intersubject and control subject comparisons show no acoustic/microwave crosstalk, either mechanical or electrical. No mode activity was measurable by any subject in the mode frequency region of any other subject, with or without zero mode pumping. [Work sponsored by Mentec, Inc.]

FRIDAY AFTERNOON, 20 NOVEMBER 1987

UM AUDITORIUM, 1:30 TO 4:05 P.M.

Session CCC. Speech Communication VIII: Cross Language and Other Studies of Perception

D. Kimbrough Oller, Chairman

Mailman Center for Child Development, University of Miami, P. O. Box 016820, Miami, Florida 33101

Chairman's Introduction—1:30

Contributed Papers

1:35

CCC1. Franco-American differences in the discrimination of DA from GA. François Santon (CNRS, Marseille, France) and Bertram Scharf (CNRS and Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

We synthesized DAs and GAs that were supposed to differ only in the third formant, between 2 and 3 kHz. Discrimination was measured in narrow-band noise maskers centered on frequencies from 1 to 3.5 kHz. The signal-to-noise ratio was set to yield nearly 100% correct responses in the least effective masker (at 3.5 kHz) and nearly 50% in the most effective masker (at 1 or 2.5 kHz). Three American listeners yielded results similar to tuning curves. Discrimination was poorest in the 2.5-kHz noise and improved rapidly at lower and higher noise frequencies. Three French listeners gave similar results, but three others gave very different results. The signal-to-noise ratios were 11 dB lower and discrimination was poorest in the 1-kHz noise. (One American also discriminated at the lower signal-to-noise ratio when given feedback.) Analysis of the tapes revealed a low-frequency click in the DAs but not in the GAs. Americans did not use this cue (without feedback). Some French also did not, but others did. Apparently, because natural American DAs differ from French DAs, French listeners are more likely to use artifactual cues.

1:50

CCC2. Perception of vowel nasalization in VC contexts: A cross-language study. Kenneth N. Stevens (Room 36-517, Massachusetts Institute of Technology, Cambridge, MA 02139), Amália Andrade, and M. Céu Viana (Centro de Linguística da Universidade de Lisboa, Av. 5 de Outubro, 85-6, 1000-Lisbon, Portugal)

A series of stimuli ranging between [tAtA], [tʌtA], and [tAnA] were synthesized by systematically manipulating (1) the duration of nasalization in the vowel [A], (2) the amount of nasalization in the vowel, and (3) the duration of the nasal murmur following the vowel. The stimuli were presented to native speakers of Portuguese, English, and French, which differ with respect to the occurrence of nasal vowels in their phonological systems. The listeners were asked to judge, for each stimulus, (1) the presence or absence of nasalization and (2) the adequacy of the stimulus as a natural utterance with respect to its nasalization. The different language groups gave similar responses with regard to the presence or absence of nasalization. However, judgments of the naturalness of the stimuli in the different languages depend on the temporal characteristics. French listeners preferred a longer duration of nasalization in the vowel and were indifferent to the presence of murmur; English listeners pre-

ferred some murmur and accepted briefer nasalization in the vowel; and Portuguese responses were intermediate. Preliminary acoustic data from utterances in the three languages are in accord with these perceptual findings.

2:05

CCC3. Acoustic-phonetic analysis of Spanish /l/. Miguelina Guirao and Enrique A. Rosso (Laboratorio de Investigaciones Sensoriales, CONICET, CC 53, 1453 Buenos Aires, Argentina)

Acoustic measurements to examine the contribution of overlapping segments—anticipatory and carryover coarticulations—to the identification of American Spanish /l/ were performed. The consonant was combined with five vowels in naturally produced CV, VC, VCV, and CCV syllables. Acoustic profiles through analysis of formant trajectories, fundamental frequency, intensity and duration of vowel, consonant, and their overlapping segments were obtained. In searching for the corresponding critical spectral features, digitalized waveforms were modified by deleting various temporal segments in the syllable. The excised sounds were used as stimuli in identifications tests. Results indicate that the duration of the syllabic nuclei varies across front, medial, and back vowels and also in pre- and post-vocalic position. Segment /l/ presented in isolation is poorly identified. Best identifications are obtained when the stimuli are integrated by the end of the overlapping portion and the beginning of the vowel. A trading off relationship effect is observed between these two segments.

2:20

CCC4. On the identification of Spanish voiced stops. Jorge A. Gurlekian, Miguelina Guirao, and Horacio Franco (Laboratorio de Investigaciones Sensoriales, CONICET, CC 53, 1453 Buenos Aires, Argentina)

Experiments were conducted to examine the interaction between relevant acoustic features of Spanish /b,d,g/ emitted with /a/ in CV syllables and VCV combinations with the first or second vowel with stress. Duration of steady and transitional vowel segments and the closure duration were systematically segmented at particular time intervals. The entire sequences and spliced segments were presented to native Spanish speakers for identification. Results were presented as a ratio between different closure durations and the number of vocalic periods required for the identification of the stop. Different ratios were obtained for each consonant in the CV syllable. When the closure portion was removed completely, the unvoiced counterparts were accurately perceived. This was not true for /g/, where the absence of the burst avoided the shifting to /k/. For the VCVs when the closure portion was entirely removed and the remaining parts

were rejoined, the stops—approximants in this context—were still identified. These results showed that /b/ and /d/ identification was cued by a voiced portion preceding the transition of the following vowel. Primary features for these sounds were located in the transition edges—final and initial—which overlapped both vowel and consonant information.

2:35

CCC5. Training new fricative contrasts. Donald G. Jamieson (Department of Communicative Disorders, University of Western Ontario, London, Ontario N6G 1H1, Canada), David Morosan (Department of Psychology, University of British Columbia, Vancouver, Canada), and Susan Rvachew (Department of Psychology, University of Calgary, Calgary, Alberta T2N 1N4, Canada)

A perceptual fading technique [cf. Jamieson and Morosan, *Percept. Psychophys.* 40, 205–215 (1986)] was used to train two types of listeners to perceive new fricative contrasts. One group consisted of young adult francophones who had difficulty producing and perceiving the voiced/voiceless “th” distinction in English. With this group, training using the fading technique improved identification accuracy both for the specific, synthetic targets used in training, and for multiple, natural tokens of these sounds, spoken by different individuals, both men and women. However, this learning was quite specific, as performance was poor when listeners were tested with the target sounds in new word positions, or when /d/ tokens were used in the distractor set. The second group of listeners were young children who consistently mispronounced at least one English fricative sound, and had been diagnosed as having a functional articulation disorder. A sizeable proportion of these children, all of whom had been selected on the basis of production difficulties, displayed atypical perceptual skills; for these children, training produced a measureable improvement in identification accuracy, and in some instances, perceptual training (without explicit production training) also improved the productions of the target sounds. [Work supported by NSERC AHFMR and HWC:NHRDP.]

2:50

CCC6. 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, Montreal, Quebec H3E 1H6, Canada)

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 were 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 sentences 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 context larger than that of the syllable. [Research done at the Université de Montréal and supported by National Research Council of Canada and Québec Government.]

3:05

CCC7. Non-native speakers can gauge degree of foreign accent in English. James Emil Flege (Department of Biocommunication, UAB, University Station, Birmingham, AL 35294)

Previous research indicated that non-native speakers can detect foreign accent in their second language (L2), even if they themselves speak it with an accent. This study used interval scaling to help determine whether

they do so by noting native-language (L1) phones in the L2, or by noting divergences from norms established for phones in the L2. English sentences spoken by Chinese and native English talkers were presented to three groups of listeners. The Chinese listeners, who spoke English with strong foreign accents, were able to distinguish Chinese from native talkers. Those experienced in English resembled native English listeners significantly more than those who were relatively inexperienced, suggesting the experienced Chinese listeners had developed tacit knowledge of how English sentences “ought” to sound. Removing pauses from the sentences did not affect foreign accent scores, suggesting that “fluency” and “foreign accent” represent different dimensions pertaining to L2 proficiency. Chinese talkers who learned English before the age of 12 years received significantly lower scores than native English talkers. Learning an L2 before 12 years—the age often associated with the end of a “critical period”—does not guarantee accent-free pronunciation. There was no difference in foreign accent scores obtained for adult Chinese learners of English who had lived in the United States for 1 and 5 years. This suggests that amount of (unaided) L2 experience may not improve L2 pronunciation, at least beyond a certain point.

3:20

CCC8. Perceptual segregation of concurrent vowels. Peter F. Assmann and Quentin Summerfield (MRC Institute of Hearing Research, University Park, Nottingham NG7 2RD, England)

The ability to group together the acoustic components of a voice and segregate them from competing voices is an important skill in speech communication. To study the auditory processes involved, pairs of vowels were presented simultaneously to listeners with normal hearing. Identification performance improved as the fundamental-frequency difference between the members of the pair increased. Two auditory models were compared to predict this result. The composite waveform was passed through an array of bandpass filters simulating the ear’s limited frequency resolution. The “place” model estimated the two fundamentals from peaks in the profile of the rms outputs of the filters. It then sampled the profile at integer multiples of each fundamental to reconstruct the spectra of the constituent vowels. The “place-time” model estimated the fundamentals from dominant periodicities in the waveforms emerging from the filters. The amount of synchronization to each fundamental was plotted as a function of filter center frequency to reconstruct the spectra. The place-time model estimated the two fundamentals more accurately than the place model, and predicted identification responses more precisely. This outcome is compatible with the idea that a temporal analysis is involved in the segregation of concurrent sounds.

3:35

CCC9. Dichotic duplex perception and modularity. Ignatius G. Mattingly (Haskins Laboratories, 270 Crown St., New Haven, CT 06511)

Recent work suggests that dichotic duplex perception (in which the same acoustic material simultaneously yields both a phonetic and a non-phonetic percept) occurs when the source-defining or source-localizing processes are given ecologically implausible information, and provides further evidence for the modularity of speech perception. If F_3 and F_1-F_2 of a synthetic syllable are presented dichotically, with identical F_0 contours, one hears just the expected syllable at some fixed position within the head. With a somewhat higher F_0 contour on F_3 than on F_1-F_2 , one hears two voices, one at each ear, both saying the syllable. But with a considerably higher F_0 contour on F_3 , perception is duplex: one hears a buzz at the F_3 ear and the syllable at the F_1-F_2 ear. And if, instead, the basic pattern is modified by shifting F_3 abruptly to the F_1-F_2 ear after 80 ms or more, one hears just the syllable, now moving across the head. But if F_3 is shifted sooner, perception is again duplex: one hears a chirp at the ear initially receiving F_3 and the syllable at the other ear. Evidently, sufficiently extreme conditions disrupt source definition or source localization without disrupting dichotic speech perception. Speech perception appears to be independent of these other modular processes, and therefore modular itself. [Work supported by NIH Grant HD-1994.]

CCC10. Synthesis and perception of paragraph-level prosody. Kim E. A. Silverman (Room 2C-440, AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Discourse structure systematically influences the prosody of utterances. In this study, units of prosodic structure that span multiple sentences (prosodic paragraphs) were synthesized by (i) temporarily increasing the overall pitch range at the start of a (textual) paragraph; (ii) gradually decreasing the range over a fixed time window at the end; and (iii) lengthening the pause at the boundary. Six ambiguous multipara-

graph texts were constructed: In each text the overall meaning (e.g., where a speaker's turn finished) depended on the location of one particular paragraph boundary. In a series of perception experiments, listeners heard one or another version of each text spoken by a formant synthesizer, and answered forced-choice comprehension questions. Overall results showed that (i) listeners do exploit such wide-domain prosodic organization when comprehending extended passages of connected speech, (ii) the structure can be signaled by F_0 alone, although (iii) the more redundantly it is encoded, the better it will be perceived. Such prosodic structure also improves the naturalness of synthetic speech. The results argue that prosody is hierarchically organized, rather than being a strictly linear sequence of intonational phrases.

FRIDAY AFTERNOON, 20 NOVEMBER 1987

UMS 4 AND 5, 1:30 TO 5:00 P.M.

Session DDD. Underwater Acoustics X: Bottom Interacting Ocean Acoustics II (Précis-Poster Session)

George V. Frisk, Chairman

Bigelow 208, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

Chairman's Introduction—1:30

Contributed Papers

Following presentation of the précis, posters will be on display until 5:00 p.m.

1:35

DDD1. Computation of the Biot coefficients. B. Yavari and A. Bedford (Department of Aerospace Engineering and Engineering Mechanics, The University of Texas, Austin, TX 78712)

To apply the Biot theory to the study of the interaction of acoustic waves with ocean sediments, the coefficients in the equations must be evaluated. The coefficients depend on the frequency and the fabric, or microstructure, of the sediments. A method for evaluating the drag and virtual mass coefficients has recently been developed [A. Bedford *et al.*, *J. Acoust. Soc. Am.* **76**, 1804–1809 (1984)]. The method requires solving for the motion of the fluid in the pores when the solid constituent is subjected to a spatially uniform, oscillatory motion. The finite element method has been used to determine the fluid motion. The drag and virtual mass coefficients have been determined for a variety of two-dimensional pore geometries. It was found that the drag coefficient is insensitive to the pore geometry, while the virtual mass coefficient is very sensitive to the pore geometry. It was determined that the coefficients are independent of the frequency up to a characteristic value of dimensionless frequency and exhibit a bilinear dependence on frequency. [Work supported by ONR.]

1:39

DDD2. The effects of seabed interaction on air-to-water sound transmission. David M. F. Chapman (Institute of Sound and Vibration Research, The University, Southampton SO9 5NH, United Kingdom)

The presence of a reflecting seabed alters the transmission of sound from an air-borne source to a water-borne receiver. Resonances in a plane-wave transmission coefficient correspond to excitation of the acoustic normal modes in the water layer. Although the air-borne source excites the same normal modes as a water-borne source, the amplitude and phase of the excitation coefficients are different. There are some notable effects: the weak dependence of transmission loss upon source height, the preferred excitation of high-order modes, and the $25 \log R$ dependence of transmission loss upon range in the mode-stripping regime. The predictions of a simple normal-mode transmission loss model are presented, including realistic absorption in the air and at the seabed. [Work supported by the U.K. Ministry of Defence, through the Royal Aircraft Establishment at Farnborough.]

1:43

DDD3. Seabed characterization from normal incidence reflectivity. Charles W. Holland and Burlie A. Brunson (Planning Systems Incorporated, 7925 Westpark Drive, McLean, VA 22102)

Surficial values of sediment compressional velocity, density, and rms interface roughness are estimated from the coherent component of normal incidence reflectivity in the frequency range 10–1000 kHz. The Kirchhoff approximation is employed for the scattering function in the inversion algorithm such that the coherent reflection coefficient is assumed to be dependent on the distribution of the bottom heights and independent of the shape of the bottom. The functional form of the reflection data suggests that the distribution of bottom heights follows a Laplacian rather than Gaussian form for a variety of sediment types. Laboratory analysis of bottom core samples provided the means for quantitative evaluation of velocity and density predictions.

1:47

DDD4. A numerical treatment of the fluid/elastic interface under range-dependent environment. Er-chang Shang (Yale University, Department of Computer Science, New Haven, CT 06520) and Ding Lee (Naval Underwater System Center, New London, CT 06320)

The actual physical problem becomes considerably difficult and complicated when a fluid medium is interfacing with an elastic medium under range-dependent environment. In this case, a set of three "continuity conditions" must hold: (1) the continuity of vertical components of displacement, (2) the continuity of vertical components of stress, and (3) the horizontal components of stress must vanish on the interface. The implicit finite difference (IFD) scheme, used to treat the fluid/fluid interface [S. T. McDaniel and D. Lee, *J. Acoust. Soc. Am.* **74**, 855 (1982)] can be extended to treat the fluid/elastic interface with some modifications. In this presentation, a numerical procedure is proposed to treat the fluid/elastic interface under a range-dependent environment. An attractive feature of this numerical model is that it is readily adaptable into the existing IFD code without excessive effort. Some computational problems will also be discussed.

DDD5. Transmission of sound and beams through a water-sediment interface. T. G. Muir and N. L. Lunde (SACLANT ASW Research Centre, I-19026 La Spezia, Italy)

The computations of Schmidt and Jensen [J. Acoust. Soc. Am. Suppl. 1 76, S10 (1984)] are extended to examine the penetration of sound beams into a sand and a mud sediment as a function of frequency, grazing angle, beam size, and pulse shape. The computations are made with the SAFARI model, an exact numerical solution to the wave equation in horizontally stratified environments. The effects of sediment attenuation and rigidity are accounted for. Beam shape contours and loss coefficients in transmission through the interface are presented. The effect of interface transmission on the integrity of propagating pulse waveforms is discussed. It is shown that decreasing the frequency to the low kHz range enhances the penetration into rigid sediments at low grazing angles (below the critical grazing angle). However, this results in a geometric dispersion effect on the transmitted waveforms, which destroys the coherence of pulsed signals.

1:55

DDD6. Bottom interacting pulse propagation in the ocean. Michael D. Collins (Department of Engineering Sciences and Applied Mathematics, Northwestern University, Evanston, IL 60201)

The parabolic equation method was first applied to underwater sound propagation by F. D. Tappert. It has since been refined and benchmarked into a valuable tool for frequency domain calculations. Development of an equivalent nonlinear method in the time domain was initiated by B. E. McDonald and W. A. Kuperman. A linear version of this method has been developed to handle bottom interaction. The new model allows density and loss to be functions of depth. Loss is accounted for in the proper manner for typical ocean sediments. Loss per meter increases linearly with frequency. A new initial field has been derived using the method of matched asymptotics. Numerical methods suited to linear acoustics have been implemented. The method has been benchmarked by convolving an impulse response obtained by it into a cw response and comparing with normal mode results. [Work supported by ONR.]

1:59

DDD7. Acoustic transmission across the water/sediment interface measured during *in situ* experiments. Lawrence J. Satkowiak (Naval Coastal System Center, Code 4120, Panama City, FL 32407-5000)

A series of acoustic measurements were made to measure the transmission of acoustic energy across the water/sediment interface. The *in situ* measurements utilized a three-dimensional array of hydrophones as acoustic receivers buried in the sediment. The acoustic sources were mounted on a movable tower located on the seafloor. The transmission was measured as a function of grazing angle, frequency, and pulse shape for two distinctly different sediment types, namely, fine sand and soft mud. The experimental results presented will be compared to predictions made using a simple multilayered model that treats the bottom as a series of contiguous layers with known acoustical properties.

2:03

DDD8. Modeling depth-dependent absorption in marine sediments. Paul R. Ogushwitz (76 Cedar Lake West, Denville, NJ 07834)

Biot model estimates of absorption are particularly sensitive to porosity and to the frame moduli. In previous work [P. R. Ogushwitz, J. Acoust. Soc. Am. 77, 429-464 (1985)], the frame moduli from a self-consistent theory of composites (SCT) was estimated, and an empirical factor (the "contact factor") that scales the SCT estimates in order to match measured wave speeds was determined. It is shown that the contact factor is related to grain packing and cementation. Here, it is first shown that Biot model estimates of sound speed versus depth do agree with measurements in Gulf Coast brine sands [G. H. F. Gardner *et al.*, Geophysics 39, 770-780 (1974)], where the SCT estimates have been scaled by the appropriate contact factor. Second, it is shown that the corresponding Biot estimates of absorption versus depth are within the range of mea-

sured absorption values, that they exhibit a maximum at 1750-m depth, and that the behavior of this maximum is consistent with other published measurements and theories. Third, Biot estimates of shear wave absorption are presented. Finally, a physical model for the contact factor is presented.

2:07

DDD9. Determination of ocean bottom sound-speed profiles using a modal inverse technique in a range-dependent shallow water environment. George V. Frisk, James F. Lynch, and Subramaniam D. Rajan (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A modal inverse technique for determining ocean bottom sound-speed profiles is applied to experimental data obtained in a range-dependent shallow water environment. The method consists of towing a cw source away from moored receivers to obtain a well-sampled measurement of the complex pressure versus range. The Hankel transform of the measured field then yields the depth-dependent Green's function versus horizontal wavenumber, which contains distinct peaks at the modal eigenvalues. The eigenvalues are used as input data to a perturbative inverse scheme that generates the sound-speed profile of the bottom. Data at 140 and 220 Hz were obtained in May 1984 in Nantucket Sound, MA in an area where the depth changed abruptly at about the midpoint of the experimental track. An associated change in bottom properties was assumed, and the concept of local modal eigenvalues yielding local geoacoustic parameters was then used to determine separate models for the two regions. Excellent agreement between the pressure fields computed for this range-dependent model using adiabatic mode theory and the experimentally measured fields is demonstrated.

2:11

DDD10. Model and data comparison for bottom backscattering. Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195) and Kevin B. Briggs (Naval Ocean Research and Development Activity, NSTL, MS 39529)

Acoustic and geoacoustic data from two sites have been compared with a model [D. R. Jackson *et al.*, J. Acoust. Soc. Am. 79, 1410-1422 (1986)] for bottom backscattering strength at high frequencies. At one site, the sediment was a fine sand with prominent, directional ripples. At the other site, the sediment was a mixture of clay, sand, and shell. The geoacoustic data were used to obtain values for the following model input parameters: sediment sound speed, sediment mass density, and the power spectrum for bottom microrelief. The model also requires a sediment volume scattering parameter, which was obtained by fitting backscattering data. The model generally gave good descriptions of the level and grazing angle dependence of the backscattering strength, and the results indicate that roughness scattering was dominant at the sandy site, while sediment volume scattering was dominant at the clayey site. At the sandy site, the microrelief spectrum showed appreciable anisotropy and an unusually rapid falloff with increasing spatial frequency. For this site, the backscattering model overestimated both the degree of anisotropy (none was observed) and the level of scattering at the highest frequencies. [Work supported by ONT and NORDA.]

2:15

DDD11. High temporal resolution observations of high-frequency acoustic bottom echoes. Thomas L. Clarke, John R. Proni (Ocean Acoustics Division, NOAA/AOML, 4301 Rickenbacker Causeway, Miami, FL 33149), and Lloyd Huff (NOAA/NOS/C&GS, 6001 Executive Boulevard, Rockville, MD 20850)

Acoustic bottom echoes at frequencies from 10 to 380 kHz were recorded for a variety of bottom types in southern Chesapeake Bay. Short pulse lengths and direct digital recording allowed temporal resolution of 100 μ s to be achieved. Supporting measurements of sediment characteristics and bottom roughness allowed comparison of the measurements with theory. The field observations were in good agreement with the theory after corrections for the interaction of transducer beam patterns with bottom roughness were made.

DDD12. High-resolution acoustic bottom roughness measurement in support of bottom echo interaction modeling. W. P. Dammann and C. A. Lauter (Ocean Acoustics Division, NOAA/AOML, 4301 Rickenbacker Causeway, Miami, FL 33149)

A high-resolution acoustic bottom profiler using an extremely narrow-beam, 3-MHz echo sounder was developed at the Ocean Acoustic Division of NOAA/AOML. The device was used to measure bottom roughness over a range of scales from less than 1 cm to several meters. Roughness measurements were made in the lower Chesapeake Bay area over mud, fine to medium grain sand, and coarse grain sand. The data produced were used to appraise the performance of an acoustic echo formation model that predicts the effects of marine bottom characteristics on a reflected acoustic pulse envelope. Major aspects of the design and use of the system, procedures for processing generated data, and examples of processed output are presented.

2:23

DDD13. Characterization of seafloor type and roughness from 12-kHz acoustic backscattering measurements. C. de Moustier (Scripps Institution of Oceanography, A-005, La Jolla, CA 92093) and D. Alexandrou (Department of Electrical Engineering, Duke University, Durham, NC 27706)

Seafloor acoustic backscatter measurements were carried out with 12-kHz sea beam multibeam echo-sounders over a variety of terrains including pelagic sediments on the top of two seamounts in the tropical North Pacific; hemipelagic sediments, lava flows, and debris flows on the East rift zone of the Kilauea volcano; and a manganese nodule field in the tropical Northeastern Pacific. These data are subjected to statistical tests for homogeneity and stationarity, and sidelobe interference cancellation is performed through adaptive filtering. These measurements are then used to compare the 12-kHz acoustic signature of these terrains based on statistics of the amplitude and the phase of the echoes received at discrete angles of incidence in the interval $\pm 20^\circ$. Implications for bottom type identification as well as surface roughness determination are discussed. [Research funded by ONR.]

2:27

DDD14. Geoacoustic propagation through random seafloor models. M. E. Dougherty and R. A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

Finite difference forward models of elastic wave propagation through laterally heterogeneous upper oceanic crust are presented. The finite difference formulation is a two-dimensional solution to the elastic wave equation for heterogeneous media and implicitly calculates P and SV propagation, compressional to shear conversion, interference effects, and interface phenomena. Random velocity perturbations with Gaussian and self-similar autocorrelation functions and different correlation lengths are presented that show different characteristics of secondary scattering. The presence of a water-solid interface in the models allows for the existence of secondary Stoneley waves, which account for much of the seafloor noise seen in the synthetic seismograms for the laterally heterogeneous models. "Random" incoherent secondary scattering increases as the product of wavenumber and correlation distance (ka) approaches 1. Deterministic secondary scattering from larger heterogeneities is the dominant effect in the models as ka increases above 1. Secondary scattering also shows up as incoherence in the primary traces of the seismograms when compared to the laterally homogeneous case. Cross-correlation analysis of the initial P -diving wave arrival shows that, in general, the correlation between traces decreases as ka approaches 1.

2:31

DDD15. Acoustic investigation of Paleo-drainage networks in central Chesapeake Bay. Jeffrey Halka (Department of Natural Resources, Maryland Geological Survey, 2300 St. Paul Street, Baltimore, MD 21218) and D. L. Gardner (NOAA, N/CGx3, Rockville, MD 20852)

Located west of the deep axial channel in the central Chesapeake Bay

is a featureless, gently sloping platform bounded by the 8- and 16-m depth contours. Sediments on this platform consist almost entirely of muds accumulated during the Holocene rise in sea level. This area has been extensively surveyed using 5- and 27-kHz seismic reflection profiling equipment. While generally revealing only minor reflectors in the muds which overlie the pre-Holocene erosion surface, certain areas consisted of acoustically impenetrable sediments. X rays of cores collected from these areas showed numerous gas-filled voids in the sediment. Stream channel margins are evident on the pre-Holocene erosion surface, and the gaseous zones are confined between these margins, although the bottom of the channels cannot be observed due to the acoustic impenetrability. The gas zones are traceable and form an interconnected network, one of which connects to a present day upland stream.

2:35

DDD16. A comparison of measured backscatter strengths with the first- and second-order terms of a Neumann series solution to the Helmholtz scattering integral. Richard Keiffer, M. F. Werby (NORDA Numerical Modeling Division, NSTL, MS 39529), and Steve Stanic (NORDA Ocean Acoustics Division, NSTL, MS 39529)

In previous ASA meetings, a rough interface scattering model under development at NORDA has been reported. This model makes use of Brekhovskikh's tangent plane approximation for the total field near the scattering surface in the Helmholtz integral equation. Direct numerical integration is then performed yielding the first-order term of a Neumann series solution for the scattered field. Furthermore, as has been shown, this model also calculates a second-order term via Neumann's successive approximation technique. Physically, this second-order approximation begins to account for secondary scatterings as well as ameliorating the initial tangent plane approximation. In this meeting, the first- and second-order modeled results with backscatter strengths recently measured off of Panama City, FL, by NORDA's Ocean Acoustics Division will be compared.

2:39

DDD17. Correcting the Kirchhoff rough boundary. Interaction model for scattering. John J. McCoy (The Catholic University of America, Civil Engineering, Washington, DC 20064)

The Kirchhoff rough boundary interaction model is an incomplete large k approximation, which does not incorporate either shadowing or multiple interaction effects. Numerical implementation suggests that, of these two shortcomings, the neglect of shadowing is the more important. Accordingly, attempts have been made at correcting the Kirchhoff model by introducing a shadow function. For statistical formulations, this introduction is frequently accomplished after the incorporation of a statistical average. This is inconsistent with the underlying physics. Here, shadow functions are shown to have a rigorous basis as statistical measures of the rough boundary geometry, independent of any scattering experiment. That is, one can define a suite of statistical measures of the rough boundary geometry, which can subsequently be interpreted as shadow functions in the context of a scattering experiment. The particular statistical measure that is appropriate for a specific experiment is determined by the statistic of the acoustic signal to be estimated.

2:43

DDD18. Sound scattering from a randomly rough fluid-solid interface. Dalcio K. Dacol and David H. Berman (Code 5160, U.S. Naval Research Laboratory, Washington, DC 20375-5000)

A general formalism for acoustic scattering from a fluid-elastic solid interface is developed. The theory parallels Waterman's work [P. C. Waterman, *J. Acoust. Soc. Am.* **45**, 1417 (1969)], based on the extinction theorem, for acoustic scattering from impenetrable surfaces. This formalism is then used to study the scattering of sound waves from a randomly rough fluid-solid interface by using perturbation theory to solve the equations for the scattering amplitudes to second order. The Watson-Keller renormalization ansatz [J. G. Watson and J. B. Keller, *J. Acoust. Soc. Am.* **74**, 1887 (1983)] is then used to account for some multiple scattering

effects beyond double scattering. A numerical study is presented comparing acoustic scattering in the fluid from a fluid–solid interface with acoustic scattering in the same fluid from a hard, impenetrable surface (Neumann boundary condition). Substantial differences between the two cases are found. Besides the expected acoustic energy loss by transmission into the solid, it was also found that the angular distribution of the remaining acoustic energy that is reflected and scattered by the fluid–solid interface differs greatly from the hard impenetrable surface case, especially near grazing incidence and scattering angles.

2:47

DDD19. Resonance interaction of sound waves with a stratified seabed. Greta Conde, P. D. Jackins, and G. Gaunard (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20903-5000)

A novel approach to analyze the reflection and transmission of sound waves from a penetrable and layered fleet interface between two media is described. This approach generalizes the first-order prediction of the resonance scattering theory (RST), which was developed earlier for stacks of arbitrarily many elastic plates separating dissimilar fluid media [Jackins and Gaunard, *J. Acoust. Soc. Am.* **80**, 1762–1776 (1986)]. The generalization consists of two portions. The first one makes the lower fluid medium become an *elastic* half-space, which makes the model useful to predict reflections from a stratified ocean bottom. This portion was briefly analyzed earlier [*J. Acoust. Soc. Am. Suppl.* **1** **80**, S116 (1986)]. The second portion, now married to the first, consists in a second-order approximation for the parameters involved in the description of the bottom reflection/transmission coefficients, which retains second-order terms containing second derivatives of the pertinent parameters. This is an application to flat, layered media of an (second-order) acoustical counterpart of an *R*-matrix formulation for many-body scattering. It is shown how the present generalization contains the earlier first-order (RST) prediction as a particular case, and how it improves its accuracy with a relatively minor additional calculational effort.

2:51

DDD20. A scattering function for bistatic reverberation calculations. Dale D. Ellis (Defence Research Establishment Atlantic, P. O. Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada) and D. R. Haller (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

The ocean bottom scattering function $S(\theta, \phi; \theta', \phi')$ depends, in general, on the grazing angles θ, θ' and azimuthal angles ϕ, ϕ' of the incident and scattered energy. However, most measurements are for backscatter only: $B(\theta) \equiv S(\theta, \phi; \theta, \phi + \pi)$. The few available general measurements indicate strong forward scattering near the specular angle ($\theta' = \theta, \phi' = \phi$), and weaker, azimuthally isotropic, diffuse scattering away from the specular. The proposed function combines Lambert's law with Brekhovskikh and Lysanov's surface scattering function, giving

$$S(\theta, \phi; \theta', \phi') = \mu \sin \theta \sin \theta' + \nu(1 + \Delta\Omega)^2 \\ \times \exp[-(1/2\sigma^2)\Delta\Omega],$$

where $\Delta\Omega = [\cos^2 \theta + \cos^2 \theta' - 2 \cos \theta \cos \theta' \cos(\phi - \phi')]/(\sin \theta + \sin \theta')^2$. By fitting μ, ν, σ to the backscattering measurements, a general scattering function is obtained. This is an improvement over two common methods of extrapolating backscattering to general scattering. In particular, one common method uses the separable approximation $\sqrt{B(\theta)B(\theta')}$, which peaks at $\theta' = \pi/2$, rather than at the specular angle. Another common method uses the half-angle approximation $B[\frac{1}{2}(\theta + \theta')]$ which peaks at the correct in-plane grazing angle (if θ' is replaced by $\pi - \theta'$ for forward scattering), but fails out of plane.

2:55

DDD21. A 3-MHz acoustic concentration meter that measures suspended sand. Daniel M. Hanes (Division of Applied Marine Physics, Rosenstiel School, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

A 3-MHz acoustic concentration meter (ACM), which measures the intensity of backscattered acoustic energy, has been deployed in the laboratory and in the field in order to estimate the concentration of suspended sand. Techniques for measuring the profile of the concentration of suspended sand close to the seabed with a vertical resolution of about 1 cm are discussed. The scattering characteristics of suspended sand in a laboratory tank were used to provide a method of calibration for the ACM. The dependence of the scattering coefficients upon sediment size was also investigated. The ACM was deployed in the surf zone at Stanhope Lane, Prince Edward Island, Canada, in October 1984. In the surf zone, the limitations of such profilometers are (a) air bubbles injected into the water by breaking waves, and (b) the possible variations in sand size with height above the seabed. [Work supported by Naval Facilities Engineering Command.]

2:59

DDD22. Modeled effects of seafloor anisotropy using the facet-ensemble method and comparison with Red Sea data. Henry F. Schreiner (Naval Ocean Research and Development Activity, Code 245, NSTL, MS 39529-5004)

Bathymetric measurements, as well as a knowledge of the formation processes involved in shaping the ocean bottom, have indicated a strong anisotropy in bottom roughness over much of the world's oceans [C. G. Fox and D. E. Hayes, *Rev. Geophys.* **23**, 1–48 (1985)]. This anisotropy can be approximated by a ridge or trough or a series of ridges and troughs. The facet-ensemble method computes the three-dimensional complex acoustic pressure from a signal that has been scattered and reflected from this kind of surface [W. A. Kinney *et al.*, *J. Acoust. Soc. Am.* **73**, 183–194 (1983)]. The model has been adapted to a side-scanning sonar of frequency 3.5 kHz and horizontal and vertical beam widths of 10° and 30° , respectively. The computation is driven by a 100-ms source and sampled at a rate of 128/s. Comparisons are made with experimental data from the highly lined Red Sea Channel. A one-to-one correspondence between the features on the echo track and the modeled results is observed. [Work supported by Office of Naval Technology.]

3:03

DDD23. Double scattering from two parallel rough surfaces using the facet-ensemble approach. Henry F. Schreiner (Naval Ocean Research and Development Activity, Code 245, NSTL, MS 39529-5004)

Many features of the ocean surface and ocean bottom are generally long crested in shape and can be approximated by thin strips or facets placed edge to edge. Frequency and time-domain (impulse) solutions for diffraction and reflection off this kind of surface have been successfully modeled [W. A. Kinney and J. G. Zornig, *J. Acoust. Soc. Am.* **77**, 1403–1408 (1985)]. The introduction of a second surface of interaction creates a significantly more complex problem. Medwin and Childs presented a Huygens wavelet interpretation for double diffraction off a pair of rigid wedges or troughs [H. Medwin *et al.*, *J. Acoust. Soc. Am.* **72**, 1005–1013 (1982)]. This interpretation is used to construct a model that computes the complex acoustic pressure after scattering off two parallel facet-constructed surfaces. The effects on the correlation between two separated receivers are presented as functions of surface roughness and bottom topography in the frequency band of 500–1000 Hz. Also, comparisons are given at different sea states of single-receiver responses from backscatter off prominent bottom features for a 3-kHz 100-ms signal. All results quantify the effects of the roughness features on signal degradation. [Work supported by Office of Naval Technology.]