PROGRAM OF

The 111th Meeting of the Acoustical Society of America

Bond Court Hotel • Cleveland, Ohio • 12–16 May 1986

TUESDAY MORNING, 13 MAY 1986 HASSLER ROOM, 9:00 TO 11:15 A.M.

Session A. Noise I: Miscellaneous Topics

Elliott H. Berger, Chairman
E-A-R Division, Cabot Corporation, 7911 Zionsville Road, Indianapolis, Indiana 46268-1650

Contributed Papers

9:00

A1. Field experience in measuring and evaluating acoustical parameters for motion picture theatre environments. William J. Cavanaugh (Cavanaugh Tocci Associates, Inc., Natick, MA 01760) and Phil D. Batton (GCC Theatres Inc., Chestnut Hill, MA 02167)

Since early 1983, the General Cinema Corporation has been engaged in a major expansion and renovation program of its movie theatre complexes throughout the United States. Furthermore, recent developments in motion picture presentation technology involving films with extended dynamic and frequency ranges have led to reconsideration of and, in some cases, adaptation of conventional criteria for evaluating the acceptability of theatres for viewing and listening to films. The results of field measurements (octave-band reverberation times, room-to-room NR's, and background sound-pressure levels) are reported for a large number of facilities including data for the often disregarded 31.5-, 63-, and 8^K-Hz bands. The considerations in obtaining adequate field data and in developing criteria, as well as practical cost effective acoustical control measures developed for new generation theatres, are reviewed.

9:15

A2. An integrated approach to office acoustics in the computer age. Hsien-Sheng (Jason) Pei (Digital Equipment Corporation, 30 Forbes Road, NR05/J2, Northboro, MA 01532)

The building environmental control system and computer/business equipment are two major noise sources in modern office buildings. This paper focuses on the technical challenges that must be addressed by the acoustical engineers in the design of office acoustics in the computer age. In several important areas (such as standards, design tools, technology development, etc.), integration is examined in some detail. The state-of-the-art of building environmental control systems and computer cooling systems will be reviewed. Technology integration potentials will be presented. A case study of integration is demonstrated. Current and future research needs are identified.

9:30

A3. Noise control of banking business equipment. S. P. Ying (Gilbert/Commonwealth, P. O. Box 1498, Reading, PA 19603)

Noise control of three kinds of business equipment, namely, document printer, currency dispenser, and endorser/encoder, was investigated. These machines which use microprocessors as the control devices were developed recently for use in banking. During the development or improvement stage, an acoustic improvement program for each kind of equipment was initiated with sound power level determination, followed by noise diagnostic tests using the coherence function technique. One of the common noise sources in the equipment was the broadband printing noise which was radiated directly from the printing head and through machine structures. The currency dispenser had gear noise which resonated with the machine structure at the gear mesh frequency. Recommendations for noise reduction included utilization of sound absorption foam inside cabinets, increase of sound transmission loss of machine cases, sound isolation for structure-borne noise, and treatments of openings. Test results of some of these improvements are presented.

9:45

A4. Reducing the aural detectability of a 30-kW motor-generator set. G. R. Garinther (U. S. Army Laboratory Command, Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005-5001)

The noise produced by motor-generator sets in Army field situations causes detectability and speech intelligibility problems. A program for computing the aural detectability distance of Army material was developed and used to guide a noise reduction effort for this generator. A field expedient solution was pursued which was developed through the testing of a number of increasingly complicated barriers and enclosures. This report describes the procedure used and shows the 1/3-octave band level, the insertion loss, and the detectability contour obtained for each configuration. The insertion loss values are compared to theory, and cooling problems associated with noise reduction of generator sets are discussed. The final enclosure configuration attenuated energy in those low frequencies which cause detection, by up to 20 dB, and reduced computed detectability from 1000 to 200 m. The use of such a program can be of assistance, for the development of Army material, and for directing noise control efforts to those frequencies which control aural detectability.

10:00

A5. Low-frequency noise in jet engine test facilities. Rollin O. Boe and Mark S. Boe (R I Corporation, P. O. Box 389, Ogden, UT 84402-0389)

Jet aircraft and engine test facilities are being designed to operate using air cooling only in the exhaust system. Many of these facilities are experiencing low-frequency noise due to aerodynamic instability. The presentation will include a discussion of low-frequency noise generation in jet engine test facilities and measured data. Data are measured down to 2 Hz since much of the noise energy is in the infrasonic frequency range.

10:15

A6. Sound transmission loss in a rectangular attenuator symmetrically lined with continuously variable flow resistance. F. B. Shenoda, R. N. Haroun (National Institute of Standards, Cairo, Egypt), and Hany Selim (IBM Cairo Scientific Center, 56 Gameat el Dowal El Arabia Street, Mohandessin, Cairo, Egypt)

The sound attenuation in a free-path rectangular duct of constant cross-sectional area was theoretically studied for the case of a symmetrically decreasing wall resistance at its both ends. The change of the wall conductance from either side was thereby taken to be parabolic. In addition to the high transmission loss it possesses, this attenuator was found to be wideband and to have small reflection coefficients at both ends, if it is inserted between two similar hard ducts. The solution of the obtained Bessel differential equation consists of a combination of modified Hankel functions of the order 1/3. The attenuator characteristic at a certain frequency was found to depend only on both the duct length and an attenuator characteristic area (A) defined as $A = S/g$, where S is the actual attenuator cross-sectional area and g is a normalized wall conductance at a distance of 1 cm from attenuator ends. Compared to the case of the homogeneous lined attenuator, the insertion of the proposed attenuator between two hard ducts was found to give a much smaller reflection coefficient at comparable values of sound transmission loss. This is due to its good matching characteristics at both ends. As an example of calculations, a duct of 1-m length having a characteristic area of 500 cm^2 was found to have an average transmission loss of about 17 dB for the frequency range up to 600 Hz. Meanwhile, the reflection coefficient was found to decrease with frequency and to have a value less than 0.1 for frequencies above 350 Hz. Experimental results were found to be in fair agreement with the theoretical results. It is worth mentioning that parabolic change of the wall conductance can be realized by means of a homogeneous sound absorbing flow resistance together with a slit which changes parabolically along the length of the attenuator. By the experimental implementation of this slit, a variable perforation area having the same dependence on length and perforation ratio of more than 35% was used. Due to the small reflection coefficient it possesses at both ends, this attenuator can be used as an exhaust silencer.

10:30

A7. Comparisons between A-weighted sound-pressure levels in the field and those measured on people or manikins. George F. Kuhn (Vibrasound Research Corporation, 2855 West Oxford Avenue, Englewood, CO 80110)

The A-weighted sound-pressure levels for 75 different industrial noise spectra were determined. Previously published sound-pressure level transformations by Kuhn and Guernsey [J. Acoust. Soc. Am. 73, 95–105 (1983)] from the field, either progressive or diffuse, to the body or head surface were applied to these noise spectra and the resulting A-weighted sound-pressure levels were calculated. For the most part, a microphone at the torso or head measures a larger A-weighted sound-pressure level than a microphone in the unobstructed sound field. The range of the resulting measurement error, due to the presence of the body or the head, is approximately -1 dB(A) to $+5$ dB(A). The magnitude of the error is sensitive to the type of sound field, the subject's orientation relative to the angle of incidence of the sound, the noise spectrum, and the microphone location. Other factors, for example, the microphone size and directivity, the distance between the microphone and the body, and the absorption of the sound by different types of clothing will also affect the measurement errors. Although these factors are implicit in some of the reported transformations, their individual, explicit effect on the measurement errors remains to be determined.

10:45

A8. The use of personal computers in the calibration of noise dosimeters. Michael A. Crivaro (U. S. Department of Labor, MSHA, 4800 Forbes Avenue, Pittsburgh, PA 15213)

Personal and handheld computers are used in the acoustical calibration laboratory of the Mine Safety and Health Administration (MSHA), Pittsburgh Health Technology Center. The laboratory was established to calibrate noise dosimeters used by MSHA coal and metal/nonmetal mine inspectors. Equipment was selected to set up new systems and to make existing systems more efficient. Analog signals from some dosimeters are measured by the computer systems and converted to useful information, while other dosimeters come equipped with standard computer interfaces. Programming of the systems was done in-house. This has two advantages. First, the programmer has constant communication with the eventual end users of the system, which results in better planning. Second, the system can be adapted immediately by anyone familiar with the program, when changes in operating procedures become desirable or necessary. In general, the systems are accurate, prevent time-consuming mistakes, and can be made to guide the user through the calibration procedure. The installation of computer controlled calibration systems has proven to be successful.

11:00

A9. The use of spreadsheet programs for noise control problems. Pranab Saha (Blachford Engineers, P. C., 1899 Orchard Lake Road, Suite 105, Pontiac, MI 48053)

Spreadsheet programs, once used as a financial/business tool, can be effectively used for solving acoustics and noise control problems using a personal computer. The spreadsheet allows one to write a pseudoprogram in terms of algebraic expressions which follows algorithms similar to our thinking process. The spreadsheet is essentially a two-dimensional matrix where entries can be made both in rows and columns. The entries, called cells, contain different types of information, such as numbers and formulas. Numbers are like expressions with fixed values. Formulas are algebraic expressions where references can be made to other parts (cells) of the program. These expressions are computed in a similar fashion as is done in a calculator but at a much faster speed. Also, the spreadsheet computer program has plenty of flexibility and versatility in many areas compared to a conventional digital computer program. One of these is the automatic updating of all results (algebraic expressions) that are affected due to the change of a value in the spreadsheet. This paper discusses the VISI-CALC® computer program spreadsheet. A worker noise exposure prediction problem is demonstrated using the VISI-CALC® program with a Hewlett–Packard desktop computer.

Session B. Physical Acoustics I: High-Temperature Acoustics

Martin B. Barmatz, Chairman
Jet Propulsion Laboratory, California Institute of Technology, 4800 Oak Grove Drive, Pasadena, California 91109

Invited Papers

9:00

B1. Probing the interior of the sun and stars with acoustic modes of oscillation. Roger K. Ulrich (Department of Astronomy, UCLA, Los Angeles, CA 90024)

For reasons which are, at present, poorly understood, the sun and presumably stars like the sun undergo continuous oscillations. The most prominant of these have been identified as normal modes which are gravity-modified acoustic resonances involving the whole sun. The spatial structure of the modes is described as the product of spherical harmonics $Y_l^m(\theta,\phi)$ and a radial eigenfunction. The value of l determines the depth of mode penetration for each value of n. The identification of the modes with l and n permits a comparison of the observed frequencies with theoretical ones and tests the adequacy of our solar models. The frequency degeneracy in m is broken by rotation, and the measurement of the fine structure of the frequencies permits a determination of the depth dependence of the solar rotation rate. Present results indicate that the sun does not deviate substantially from a condition of uniform rotation. These results appear to rule out the hypothesis that the solar gravitational potential could have a large enough quadrapole term due to solar internal rotation to invalidate tests of general relativity.

9:30

B2. Acoustic pressure measurement at high temperatures. J. R. Mahan (Mechanical Engineering Department, Viriginia Polytechnic Institute and State University, Blacksburg, VA 24061)

It is often necessary to measure the acoustic pressure in a high-temperature environment as, for example, in the study of combustion noise. High-temperature microphones have been developed, and techniques have been devised for adapting "room temperature" microphones for use in the high-temperature environment. The unique features of this measurement problem are identified, and state-of-the-art methods for its solution are presented and discussed. Particular attention is paid to the acoustic waveguide probe system, in which a common room temperature microphone is thermally isolated from the high-temperature environment through a pressure transmitting tube. Results of a recent analysis of such a system are presented and compared with new measurements. It is suggested that the probe system, when used with the model to correct for axial temperature gradients along the probe, may be the optimum solution to this difficult measurement problem.

10:00

B3. Liquid metal thermoacoustic engine. G. W. Swift, A. Migliori, and J. C. Wheatley (Condensed Matter & Thermal Physics Group, Los Alamos National Laboratory, Los Alamos, NM 87545)

A liquid metal thermoacoustic engine is studied both theoretically and experimentally. This type of engine promises to produce large quantities of electrical energy from heat at modest efficiency with no moving parts except the acoustic oscillations in the liquid metal. In the engine, heat flow from a high-temperature source to a low-temperature sink amplifies a standing acoustic wave in liquid sodium. This acoustic power is simply converted to electric power by means of a magnetohydrodynamic effect at the acoustic oscillation frequency. A detailed thermoacoustic theory applicable to this engine is developed, and it is found that a reasonably designed liquid sodium engine operating between 700 °C and 100 °C should generate about 60 W/cm^2 of acoustic power at about 1/3 of Carnot's efficiency. Construction of a 3000-W thermal laboratory model engine is almost complete. A 1-kW, 1-kHz liquid sodium magnetohydrodynamic transducer has also been designed and built. It is now very well characterized both experimentally and theoretically. The first generator of its kind, it already converts acoustic power to electric power with 40% efficiency.

Contributed Papers

10:30

B4. Elastic constants of steel as a function of heat treatment. W. T. Yost (NASA Langley Research Center, Mail Stop 231, Hampton, VA 23665)

The velocity of sound and the nonlinearity parameters of HY80 3.25% NiCrMoV steel samples which were heat treated at 538 °C for periods ranging from 0–100 h were measured. Elastic constants (c_{11} and c_{111}) associated with these measurements were reported, and they were compared with values obtained from other steels. It was found that the difference between measured values among the samples is larger than one would expect. Possible reasons for the discrepancies are discussed.

10:45

B5. Resonant cavity techniques for accurate measurements of the ratio of the speed of sound to the speed of light. James B. Mehl (Department of Physics, University of Delaware, Newark, DE 19716) and Michael R. Moldover (Thermophysics Division, National Bureau of Standards, Gaithersburg, MD 20899)

In principle, measurements of the resonance frequencies of both the acoustic and the microwave modes of a single cavity can determine the ratio of the speed of sound u of a monatomic gas to the speed of light c. Such measurements, carried out with high accuracy, could determine the universal gas constant R and the thermodynamic temperature T with unprecedented accuracy. The realization of these possibilities can be greatly facilitated by judicious choices of cavity geometry and resonance modes. The present state of the art suggests that the ratio u/c can be measured to parts per million accuracy using cavities whose geometry is known only to parts per thousand. Recent experimental and theoretical results will be presented.

11:00

B6. Levitation using intense acoustic fields at high temperatures. Roy R. Whymark, Charles A. Rey, Thomas J. Danley, Gregory Hammarlund, and Dennis R. Merkley (Intersonics, Inc., 3453 Commercial Avenue, Northbrook, IL 60062)

Intense acoustic fields can be used to position objects without mechanical contact. This phenomenon finds application in high-temperature materials research by enabling a specimen to be heated, melted, reacted, cooled, and solidified in a containerless state. The acoustic force vectors capable of levitating an object are given by the gradient of the acoustic potential energy density. By suitable shaping of the acoustic field, a closed energy well can be created and a small specimen captured therein. Some recent experiments were carried out in microgravity aboard the Space Shuttle in order to reduce the requirements for intense acoustic fields, typically from 145 to 165 dB, at 15 kHz. Preliminary results are presented showing successful containerless processing of glass specimens of density 5 g/cm^3 at 1550 °C. Ground-based experiments have levitated densities of 20 g/cm^3 at STP and densities up to 4 g/cm^3 at 1000 °C. Various results from these experiments will be presented along with a discussion of the effects of acoustic cooling, thermal perturbations of the acoustic field, and the presence of harmonics. [Work supported by NASA.]

11:15

B7. Acoustic radiation force on a particle in a temperature gradient. P. Collas (Department of Physics and Astronomy, California State University, Northridge, Northridge, CA 91330) and M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

An expression for the acoustic radiation force on a small spherical particle of radius R in a standing wave field is derived. The particle is inside a long tube chamber with a temperature gradient along the axis of symmetry. Assuming $R \ll \lambda$, and neglecting convection, acoustic streaming, heat conduction, and viscosity effects, an expression for the force that consists of a "local" version of Gor'kov's result [L. P. Gor'kov, Sov. Phys. Dokl. **6**, 773 (1962)] as well as correction terms of order $\beta\lambda$, where $\beta = T'/T$, is obtained. Also the effect of various temperature gradients on the acoustic positioning (levitation) properties of the system is investigated numerically. The results of this analysis will be compared to the uniform temperature case. [Work supported by NASA.]

11:30

B8. Acoustic radiation force in a dual-temperature resonant chamber. J. Robey and E. Trinh (Jet Propulsion Laboratory, California Institute of Technology, 4800 Oak Grove Drive, Pasadena, CA 91109)

The acoustic radiation force was measured for a dual-temperature resonant chamber. This rectangular chamber has its long dimension approximately 8.5 times the square cross-sectional dimension and the opposite ends are at widely different temperatures. Force profiles were obtained for two hot-end temperatures of 500 °C and 750 °C, while the cool end remained at approximately room temperature. The lateral force was measured as a function of the long dimension of the chamber along the temperature gradient, and as a function of the drive voltage. The highest force per unit mass measured at 500 °C was 103.8 dyn/g for a frequency of 4450 Hz, and at 750 °C the highest force was 82.8 dyn/g for a frequency of 5500 Hz. Qualitatively, the measured force profiles correspond well with the theoretical curves; however, correlation of absolute magnitude is yet to be determined. [Work supported by NASA.]

11:45

B9. Application of acoustic levitation to the investigation of melting and freezing phenomena. Eugene H. Trinh (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

Ultrasonic levitators have been used in ground-based laboratories to observe melting and freezing phenomena characteristic of organic materials and metals. The behavior of freely suspended substances undergoing a first-order phase transition and the influence of a high-intensity acoustic field have been of primary interest. The undercooling ability of the materials and the acoustic field–melt interaction are related subjects, and some experimental observations have been obtained in order to elucidate this relationship. The size of the levitated samples is on the order of 1 mm and the temperature range studied is between −25° and 400 °C. [Work supported by NASA.]

TUESDAY MORNING, 13 MAY 1986 — WEST BALLROOM, 8:30 TO 11:30 A.M.

Session C. Physiological Acoustics I and Psychological Acoustics I: Otoacoustic Emissions and Evoked Response

Edward J. Walsh, Chairman

Department of Surgery, Division of Otolaryngology, Southern Illinois University School of Medicine, 801 North Rutledge Street, Springfield, Illinois 62702

Contributed Papers

8:30

C1. Low noise microphone for cochlear emissions. Mead C. Killion and Jonathan K. Stewart (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007)

A specialized microphone for use in measuring cochlear emissions contains four modified Knowles microphones mounted in mechanical opposition for minimum vibration sensitivity in their summed output. That output is electrically equalized to provide a flat frequency response from 100 Hz to 12 kHz when measured with a disposable foam eartip containing 10 mm of 3.8-mm-i.d. probe tubing. To facilitate measurement of stimulated emissions, and especially cochlear distortion products, the construction includes two 1.35-mm-i.d. earphone coupling tubes mounted to pass through the 3.8-mm-i.d. probe tubing, permitting the delivery of stimuli from two independent ER-2 insert earphones, each providing a flat eardrum-pressure frequency response to 12 kHz. The complete microphone exhibits a typical noise spectrum level of − 20 dB SPL at 1 kHz decreasing to − 26 dB at 5 kHz, 8 to 10 dB below that of a single Knowles EA-1954 microphone and 5 to 10 dB below the apparent noise level of good young ears based on the estimate of Killion [J. Acoust. Soc. Am. **59**, 424–433 (1976)].

8:45

C2. Synchronization of spontaneous otoacoustic emissions and driven limit-cycle oscillators. Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907), Arnold Tubis, and Kenneth Jones (Department of Physics, Purdue University, West Lafayette, IN 47907)

Spontaneous otoacoustic emissions (SOAE's) can be synchronized by external tones within a narrow frequency band characterized by a synchronization tuning curve [E. Zwicker and E. Schloth, J. Acoust. Soc. Am. **75**, 1148–1154 (1984)]. Surrounding this range of synchronization is a region of unstable partial synchronization leading to amplitude fluctuations of the stimuli. Psychophysical synchronization threshold curves [D. T. Kemp, Scand. Audiol. Suppl. **9**, 35–47 (1979)], based on an abrupt change in the percept of low-level stimuli near threshold microstructure, show similar tuning. These two types of synchronization tuning curves are compared in subjects with strong SOAE's and associated threshold microstructure. Simple models for these tuning curves are discussed in terms of driven limit-cycle oscillators. [Work supported by NIH.]

9:00

C3. Effects of intense sound exposure on spontaneous otoacoustic emissions. Susan J. Norton (Hearing and Speech Department, University of Kansas Medical Center, Kansas City, KS 66105), Craig A. Champlin, and John B. Mott (Speech–Language–Hearing, University of Kansas, Lawrence, KS 66044)

The effects of intense sound exposure on spontaneous otoacoustic emissions (SOAE) were measured in normally hearing human subjects as a function of exposure duration and frequency region. Consistent with the TTS literature, exposures one-half to one-quarter oct below the SOAE frequency have the largest effect. A 30-s, 105-dB SPL exposure shifts the SOAE frequency downward, while having little effect on SOAE amplitude. Recovery is biphasic: In the first minute post-exposure, the SOAE frequency changes rapidly; over the next 20 to 30 min, the SOAE slowly returns to pre-exposure values. Exposures of 15 and 60 s have similar effects. These results suggest that alterations in cochlear mechanics can be produced by stimuli that do not typically produce TTS. [Work supported by NINCDS grants R15 NS23202-01 (SJN & CAC) and T-32-NS07257 (JBM).]

9:15

C4. Auditory nonlinearities: A new approach in light of an active model. J. C. Caerou, J. M. Dolmazon, and V. S. Shuplakov (Institut de la Communication Parlée, I.N.P.G./E.N.S.E.R.G., 46 Avenue Félix Viallet, 38031 Grenoble Cedex, France)

A new model of ear which simulates the nonlinear behavior of the peripheral auditory system for real input sounds is described. The main assumption underlying the structure of the model is the presence of an active mechanism which works mainly near the resonance. The general structure of the model looks like classical lumped transmission line. Each cell has its own frequency resonance according to the coordinate-frequency correspondence observed along the basilar membrane. A nonlinear local feedback from the resistive element is mixed to the input signal in each cell. Such a circuit exhibits interesting results concerning, mainly, the evolution of selectivity with respect to the input signal level, two-tone suppression effects, and combination tones generation. The first experiments carried out by a computer simulation which uses a classical numerical method for resolution of a nonlinear differential equation system are presented. Tuning curves which show a decreasing Q factor when increasing the input level, and the results obtained with multifrequency component input signals, are presented.

9:30

C5. Latency of otoacoustic emissions and ABR wave V using tone-burst stimuli. Stephen T. Neely (Boys Town National Institute, 555 North 30 Street, Omaha, NB 68131), Susan J. Norton (Hearing and Speech Department, University of Kansas Medical School, Kansas City, KS 66105), Michael P. Gorga, and Walt Jesteadt (Boys Town National Institute, 555 North 30 Street, Omaha, NB 68131)

Auditory brain stem responses (ABR) and otoacoustic emissions (OAE) were measured in normal-hearing subjects, using tone-burst stimuli. The measured latency of wave V can be characterized as the sum of two components: (1) a "mechanical" component which varies with both intensity and frequency; and (2) a "neural" component which is independent of intensity and frequency. The mechanical component can be viewed as the time it takes for the stimulus to reach its point of maximum displacement on the basilar membrane. The neural component can be thought of as the time between this mechanical event and the generation of wave V. The determinants of OAE latencies are not completely understood. Current theories suggest that OAE's are determined by cochlear mechanical properties and that their latencies should be twice the "forward" travel time to the cochlear place which responds best to a given frequency component. Using similar stimuli, the latency of the "mechanical" component of the ABR was compared to the latency of the OAE. The data are consistent with the "two-way" travel hypothesis. [Work supported by NINCDS.]

9:45

C6. Individual differences in auditory-evoked responses: Second report. Judith L. Lauter and Robert L. Loomis (Department of Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721 and Department of Otolaryngology, Washington University, St. Louis, MO 63110)

Last spring we reported to this meeting findings from a series of auditory-evoked response (AER) tests designed for studying individual differences in AER's by comparing between-subject (BS) variability with within-subject (WS) variability. The series involved a repeated-measures design, where seven young adult subjects were tested weekly for eight sessions; in each session, AER's were collected for right monaural, left monaural, and binaural click stimulation. Continuing analysis of the brain stem data, focusing on comparisons of binaural versus monaural responses, shows further examples of: (1) clear differences between the two types of variability; (2) distinct patterns, or "profiles," of individual performance that are articulated both in terms of differences according to ear of stimulation and peak of the response; (3) striking degree of replicability of individual patterns of performance; (4) deviations from replicability that most often take one of two forms—(a) increases in stability with continued testing, and/or (b) "emergence" of response patterns that may result in the response profile of one subject coming to resemble that shown by other individuals. [Work supported, in part, by AFOSR 84-0335 and 85-0379.]

10:00

C7. The complex event-related potential (CERP): Perturbations in the high-rates auditory steady-state response following an omitted stimulus. Scott Makeig and Robert Galambos (Children's Hospital Research Center, 8001 Frost Street, San Diego, CA 92123)

When a train of brief or impulsive sound stimuli is delivered to the adult auditory system at repetition rates near 40/s, a near-sinusoidal so-called high-rates steady-state evoked response (HRR) is recorded from the scalp [Galambos *et al.*, Proc. Natl. Acad. Sci. **78**, 2643–2647 (1981)]. Amplitude and phase perturbation in this ongoing response following another experimental event can be observed by averaging responses time locked to the event and narrow-band filtering at the stimulus rate. A *complex* event-related potential (CERP) is the result. The CERP is a two-dimensional frequency-domain analog of the usual one-dimensional time-domain event-related potential (ERP). We report here a demonstration that the HRR may be perturbed for several hundred milliseconds after a single click has been dropped out of the stimulus train. The high-rates CERP provides another window on event-related brain dynamics parallel to the auditory slow-wave sequence. [Work supported by the NIH.]

10:15

C8. Human evoked potentials (EP's) reveal central masking events. Robert Galambos and Scott Makeig (Children's Hospital Research Center, 8001 Frost Street, San Diego, CA 93123)

Physiological explanations of human masking usually begin and end with hypothetical mechanical interactions of Bekesy waves at the basilar membrane level, although the coexistence of central neural events has often been postulated. We have simultaneously extracted two scalp-recorded EP's—the auditory brain stem (ABR) and the 40-Hz steady-state (SSR) responses—during stimulation with white noise and monaural clicks of fixed intensity. In the ipsilateral, or direct masking situation, the amplitude of ABR wave V (latency about 6 ms) and that of the SSR (latency perhaps 35–50 ms) both decline as the noise level increases, but with somewhat different slopes. When exactly the same noise sequence is delivered to the contralateral ear, ABR wave V is unchanged, whereas the SSR progressively declines to around 50% of its control size. The results indicate that during masking the neural activity initiated by either ipsi- or contralateral noise interacts centrally with any stimulus-related neural activity delivered to the brain by the auditory nerve. [Work supported by NIH.]

10:30

C9. Evoked potential asymmetries and behavioral responses to consonant–vowel stimuli presented dichotically and diotically. Carol A. Sammeth and S. Joseph Barry (Department of Communication Disorders, The University of Oklahoma Health Sciences Center, P. O. Box 26901, Oklahoma City, OK 73190)

Cortical auditory-evoked potentials (AEP's) were recorded simultaneously with the collection of behavioral responses to consonant–vowel stimuli presented both dichotically and diotically in a group of 16 right-handed normal females. During dichotic listening, the mean N1–P2 component of the AEP's recorded over the temporal region of the left hemisphere was found to be significantly larger in amplitude than the mean recorded over the homologous area of the right hemisphere. No significant amplitude differences were found for any AEP component during diotic listening. The latencies of AEP components revealed no systematic trend across hemispheres during either dichotic or diotic listening. The mean amplitude of N1, and the mean peak-to-peak amplitudes of P1 and N1 and N1–P2 were significantly larger over the right hemisphere during dichotic listening than during diotic listening, a result opposite to that expected if suppression of right hemisphere activity had occurred during dichotic testing. Correlation analyses revealed no significant relationships between hemispheric differences in AEP's, a measure of strength of handedness, and a measure of right-ear advantage during dichotic listening.

10:45

C10. The measurement of auditory brain stem responses in the presence of high-pass and notch noise maskers. Beth A. Prieve and Paul J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

The auditory brain stem response (ABR) was recorded from hearing-impaired persons having steeply sloping, high-frequency, sensorineural hearing loss and normal-hearing persons. Tone bursts and clicks were presented to all subjects in quiet and in the presence of either a high-passed or a 1-oct, notched-noise masker. The maskers were high passed/centered at 2 and 4 kHz. For normal-hearing subjects, the latency of wave V was greater for responses to masked stimuli than for unmasked stimuli. An increase in wave V latency was also seen for masked tone bursts or to clicks presented in the 2-kHz passed/centered masker in the hearing-impaired subjects. When clicks were presented to the hearing-impaired subjects in the presence of a high-pass masker with a 4-kHz cutoff, wave V latency increased. However, when presented in the presence of a notch-noise masker centered at 4 kHz, latency decreased. These findings suggest that the response to clicks in impaired ears may arise from neurons more apical than those in unimpaired ears.

11:00

C11. Effects of contralateral stimulation on auditory-evoked potentials. Stanley Zerlin (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

The effects of contralateral stimulation on wave V amplitude of the auditory brain stem response (ABR) over a range of interaural intensities were studied. Wave V amplitude of a "reference" ear was elicited by 4-kHz tone bursts (10/s) at 85 dB SPL; simultaneous, contralateral, 4-kHz tone bursts were presented at levels of 85, 75, 65, 55, and 45 dB SPL in addition to a no-stimulus condition. (Interaural relations were thus 0, − 10, − 20, − 30, and − 40 dB and a monaural condition.) Resulting wave V amplitudes measured on the reference side were expressed as ratios with respect to the amplitude of the monaural condition. Ratios greater than 1.0 indicated enhancement and ratios less than 1.0 indicated suppression. The 0-dB interaural condition gave a ratio of 1.5–2.0; the − 10-dB condition yielded a ratio greater than 1.0, while the − 20-dB condition gave a ratio of approximately 1.0. The − 30 and − 40-dB conditions yielded ratios well below 1.0, suggesting suppression effects at the brain stem level.

11:15

C12. Rate and frequency effects on brain stem binaural interaction responses. T. K. Parthasarathy and G. Moushegian (Communication Disorders, Callier Center, University of Texas at Dallas, Dallas, TX 75235)

Spectrum and presentation rate were parameters utilized to evaluate the binaural interaction component (BIC) of the auditory brain stem response (ABR) at 100 dB SPL in normal hearing adults. Binaural brain stem waveforms were similar in morphology to the sums of the monaurals. For stimulus conditions of this study, amplitudes of the binaural wave V were always smaller with shorter latencies (0.1–0.2 ms) than the summed monaural amplitudes and latencies. Increasing stimulus rate produced small increases in response latencies and decreases in peak amplitudes of the N1–P2 binaural interaction components and concomitantly, an increased latency and decreased amplitude of wave V of ABR. Rate effects at different center frequencies affected the BIC in ways significantly different from known effects of repetition rate on wave V ABR latencies using unfiltered clicks. Both the wave V of ABR and N1–P2 component of BIC from all subjects consistently evoked shorter latencies to the 2000-Hz tone bursts than to either 500- or 1000-Hz tone bursts. At low frequencies, a *derived* frequency following response (FFR) was obtained having latencies and configurations which suggest that its emanation is central, not peripheral or artifactual.

Session D. Speech Communication I: Perception: Various Topics (Poster Session)

Robert J. Porter, Chairman

Department of Psychology, University of New Orleans, New Orleans, Louisiana 70148

All posters will be displayed from 8:30–11:30 A.M. To allow all contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30–10:00 A.M. and contributors of even-numbered papers will be at their posters from 10:00–11:30 A.M.

Contributed Papers

D1. Predicting the future from speech timing: Japanese data. Robert Port (Department of Linguistics, Indiana University, Bloomington, IN 47405) and Daniel P. Maki (Department of Mathematics, Indiana University, Bloomington, IN 47405)

Attempts were made to simulate aspects of the perceptual system by predicting when signficant phonetic events are going to occur. The prototype model maintains a grammar of upcoming syllables and periodically computes (1) an instantaneous probability for each possible upcoming segment type using temporal information present at that point, and (2) a probability density function of the onset of prominent acoustic boundaries. Applied to some Japanese data [Port *et al.*, J. Acoust. Soc. Am. (1982)] the model was trained on the speech of five speakers who read a controlled set of Japanese words (like *baku*, *baaku*, *bakku*). It trains by finding the best linear combination of segmental durations to predict (1) locations of future segmental boundaries, as well as (2) syllable-type possibilities (given the tiny implicit grammar of the experiment). The system was tested on five different speakers reading the same list and was able to anticipate word identity with fair accuracy. Although this system relies on the mora-timing constraint in order to work (since this regularity makes future events predictable), it does not directly employ the mora in its computation.

D2. Consonant intelligibility in sentences: Effects of word position and narrow-band processing. A. Schmidt-Nielsen (Naval Research Laboratory, Code 7520, Washington, DC 20375)

The intelligibility of pairs of consonant sounds was investigated using pairs of interchangeable sentences, e.g.: *He is studying (Job /Jove) in his world religion course.* There were three pairs of sentences for each contrast, with the contrast occurring in word initial, word medial, and word final position. The test conditions included a 2400 bit/s linear predictive coding (LPC) voice processing algorithm and unprocessed speech. There were 60 listeners in each condition. The consonant pairs selected for this experiment were among those that had been shown to be most vulnerable under LPC processing in previous research [A. Schmidt-Nielsen, J. Acoust. Soc. Am. **74**, 726–738 (1983)]. Individual pairs were more poorly recognized in some word positions than in others. Contrary to previous results, the average performance for phonemes in word medial and final positions was as good as for word initial position. These results will be compared with earlier results using VCV fragments excised from running speech and with results from standard intelligibility tests.

D3. Temporal invariance for word identification. W. Reilly, R. Port (Department of Linguistics, Indiana University, Bloomington, IN 47405), D. Maki (Department of Mathematics, Indiana University, Bloomington, IN 47405), and G. Dorffner (Department of Computer Science, Indiana University, Bloomington, IN 47405)

Attempts were made to learn how well timing alone could be used to differentiate words from each other when speakers and tempo vary. In addition, we wanted to determine if this information is being missed by some commercial speech recognizers. A set of 12 phrases and words was selected describable in terms of the same sequence of acoustic intervals but differing in stress, consonant voicing, vowel tensity, etc.: *accuse, Otis, advise, it froze, outhouse, appease, abbeys*, etc.). The lists were recorded at two tempos by four speakers and prominent intervals were measured. First, discriminant analysis was employed on the measurements to identify the words. Second, the recordings were used with a commercial isolated-word speaker-dependent recognizer for training and testing. Many simple experiments were conducted with these two systems comparing their performance with training on (1) one tempo, testing on the other, (2) one speaker, testing another, and (3) one versus several speakers. The results show that (1) a temporal description for each word *can* be reliably made that is invariant across speakers and speaking tempo, and (2) for these words, timing performs about as well as the commercial system but made different errors, thus suggesting that such timing information might improve recognizer performance.

D4. The perception of preaspirated stops. Ailbhe Ni Chasaide (Centre for Language and Communication Studies, Trinity College, Dublin 2 Ireland)

Production data for preaspirated stops in Scottish–Gaelic, Icelandic, and Irish show a slow breathy-voiced transition (BVT) from the vowel to the truly voiceless aspiration (H). The experiment reported here examined (a) the perceptual relevance of the BVT, and (b) the effect that the amplitude of H has on the percept of a preaspirated stop. From computer-edited natural speech, stimuli were prepared with and without the BVT, and with differing durations of H at three amplitude levels. Results suggest that: (a) the BVT alone is sufficient to cue the preaspirated stop, even when H is entirely absent; for stimuli which did not contain the BVT, listeners fell into two groups and 40% of subjects did not judge the stops as preaspirated, even when H was long and of high amplitude. (b) For the 60% of subjects who "accepted" stimuli without a BVT as preaspirated stops, there was a strong trading relationship between the amplitude of H and its duration.

D5. The perception of voice quality: Multidimensional scaling evidence. Jody Kreiman (Phonetics Lab, Department of Linguistics, UCLA, Los Angeles, CA 90024) and George Papçun (Computer User Services C-10, Mail Stop B-296, Los Alamos National Lab, Los Alamos, NM 87545)

Forty-four listeners aged 21–74 rated the similarity of all possible pairs of ten male voices. Three three-way multidimensional scaling analyses were performed, including all listeners, listeners aged 21–44, and listeners aged 45 or older. A six-dimensional solution was selected for the group data ($r^2 = 0.655$); the dimensions were interpreted as "masculinity," "sincerity," "harshness," "variability," "breathiness," and "liveliness." Although significant differences in discrimination ability have been observed for the two age groups [Kreiman and Papçun, J. Acoust. Soc. Am. Suppl. 1 **77**, S9 (1985)], individual solutions provided no evidence for differences in perceptual strategies. The distance between each pair of stimuli on each dimension was then compared with the number of times those stimuli were confused. Differences on the first dimension correlated significantly with confusions ($r = -0.74$), but no other single dimension was significantly associated with confusability. Moreover, the total distance between stimuli (summed across all dimensions) was no better correlated with confusability than was distance on the first dimension alone. It is hypothesized that listeners evaluate voice quality using a small number of general features that apply to all voices in a population, and a relatively large number of local features that are specific to individual voices.

D6. Preliminary observations on the production and perception of clapping. Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Clapping is a rhythmic human activity that communicates approval and, possibly, individual characteristics such as size, sex, and personality. It seems to have been studied little, even though there are interesting questions to be asked about motor control, individual differences, development, and perception. In this pilot study, ten male and ten female subjects were recorded clapping at their most comfortable rate. Spectral and temporal measures were obtained, and the same subjects (all known to each other) participated in a "clapper identification" experiment. While absolute identification was predictably poor, subjects' responses were quite systematic, mainly due to a tendency to associated slower rates of clapping with male individuals. Males, in fact, clapped somewhat slower than females, whereas there was much systematic spectral variability unrelated to sex. Several principal dimensions of spectral variation were derived by factor analysis, and an attempt was made to relate them to parameters of hand configuration.

D7. Familiar voice recognition and unfamiliar voice discrimination are independent and unordered abilities. Diana Van Lancker and Jody Kreiman (Phonetic Lab, Department of Linguistics, UCLA, Los Angeles, CA 90024)

One-hundred and ten normal and thirty-two brain-damaged (BD) subjects were tested on voice discrimination and voice recognition protocols. The recognition task included samples of famous male voices in a multiple-choice format; the discrimination test consisted of pairs of male voices in a same–different task. Three findings suggest a dissociation between the two abilities. (1) Scores on the two tasks were only slightly correlated in normal subjects and were not significantly correlated in BD subjects, suggesting no obligatory relationship between performance on one task and performance on the other. (2) RBD subjects performed significantly less well than normal controls in recognition, while LBD subjects did not differ from normals; brain damage in either hemisphere affected discrimination abilities. (3) Nearly half of the BD subjects showed very large discrepancies between scores on the tasks, with chance performance on one alongside normal scores on the other. These asymmetries in performance occurred in both directions, indicating that discrimination and recognition are independent, unordered processes. Examination of CAT scans suggests that voice recognition is impaired only in right-posterior BD, while voice discrimination is impaired in damage to the temporal lobes on either side.

D8. Effects of sentence context on identification of whispered words. Jola Jakimik (Department of Psychology, University of Wisconsin–Madison, Madison, WI 53706)

In a previous study [R. Sturm and J. Jakimik, J. Acoust. Soc. Am. Suppl. 1 76, S29 (1984)], positive and negative effects of linguistic context on identification of whispered words were found. Relative to neutral contexts, appropriately biased contexts improved identification, whereas inappropriately biased contexts reduced correct identifications, and led to misperceptions, even for words accurately perceived in neutral contexts. The present study follows up on these results using more carefully controlled stimulus materials, and more sensitive response measures. Acoustically identical whispered words were presented in various contexts, and listeners rated their confidence in the words they identified. The accuracy and confidence results are examined to establish the scope of the effects of context. The implications of the findings for models of the integration of acoustic-phonetic and higher-level information in speech perception are discussed.

D9. Influence of fundamental frequency in the perception of vowel height. Maria-Gabriella Di Benedetto[a] (Research Laboratory of Electronics, Room 36-529, Massachusetts Institute of Technology, Cambridge, MA 02139)

Acoustic analysis of the vocalic portions of CVC nonsense syllables spoken by three speakers (two males and one female) shows that, in the dimension representing vowel height, individual differences for low vowels are reduced when the vowels are represented by the difference $F1$–$F0$ rather than by F. For high and mid vowels, on the other hand, a smaller shift in the $F1$ dimension would be needed to correct the differences in $F0$. Vowel identification experiments using CVC synthetic stimuli show that an increase in $F0$ from 120 to 180 Hz does not result in a clear effect on the identification functions, while a variation from 120 to 240 Hz does result in consistently different judgments. In a second experiment, one formant stimuli with $F0 = 120$ Hz and various values of $F1$ (300, 350, 400, 500, 600 Hz) were matched against one-formant stimuli in which $F1$ was adjustable and $F0$ equal to 180 or 240 Hz. The value of $F1$ for best match was usually between an exact formant match, and a match yielding similar values of $F1$–$F0$ for comparison and standard stimuli. The match was close to $F1$ for low $F1$ and $F0 = 180$ Hz, and approached (but did not reach) similar $F1$–$F0$ for higher $F1$ and $F0$ values. [a] On leave from Department of Information and Communication, University of Rome, La Sapienza, Italy.

D10. Matching of "physical" and "perceptual" spaces for vowels. Stephen A. Zahorian and Amir J. Jagharghi (Department of Electrical Engineering, Old Dominion University, Norfolk, VA 23508)

An algorithm for matching physical and perceptual spaces for psychological stimuli will be described. Target points for each stimulus class must be chosen in a multidimensional perceptual space. The physical space consists of a multidimensional measurement space, in which measurements are made of each stimulus for a large number of subjects. A linear transformation from the measurement space to the perceptual space is determined such that the mean square distance between target points and transformed measurement points is minimized. There is no requirement that the dimensionality of the measurement and perceptual spaces be the same. Thus the algorithm can be used to redefine the measurement space with fewer dimensions such that the correspondence with predefined stimulus categories is maximized. This procedure has been tested using vowels spoken in an /hVd/ context, six principal components for measurement parameters, and a three-dimensional perceptual space. Target positions in the perceptual space were based on published data from multidimensional scaling experiments for vowels. The resultant transformation has been used to map vowels to colors for use in a speech training aid for the hearing impaired. Experimental results will be given. [Work supported by the The Whitaker Foundation.]

D11. Perceptual categorization of a compressed vowel space. Douglas Varley, Karen Landahl (Department of Linguistics, University of Chicago, Chicago, IL 60637), and Herbert Jay Gould (Center of Craniofacial Anomalies, University of Illinois at Chicago, Chicago, IL 60637)

Acoustic properties of vowel tokens produced by a speaker with radical malformation of the vocal tract have been investigated. A study of vowel formant structures demonstrated that the speaker does not maintain expected intravowel formant relationships which results in a diminished vowel space. However, the speaker's vowel categories do display formant structures which are patterned after the norm [Peterson-Falzone and Landahl, *Speech and Language* (Academic, New York, 1981), Vol. 6]. The question remains in what way the anomalous formant structures will affect vowel categorization by listeners. A previous report of modeled speech [Landahl and Gould, J. Acoust. Soc. Am. Suppl. 1 77, S100 (1985)] suggests that this speaker's perceived vowel space will not include vowel categories at the periphery. The current study reports perceptual testing of vowel tokens produced by this speaker. The results of these tests concerning the existence of peripheral vowels are in accord with earlier predictions that the speaker would be unable to maintain category distinctions among more central vowels, all vowel categories being compressed in the diminished vowel space. We predicted [ɪ]→[ɛ], [æ]→[ɛ], and [ʊ]→[ɔ] mismatches. The results are essentially in line with these predictions. [Work supported by NINCDS.]

D12. Discrimination of synthetic-vowel formant amplitude change in $F2$, $F3$, with two pyschophysical methods. Roy W. Gengel and James L. Hieronymus (Institute for Computer Sciences and Technology, Building 225, Room A216, Gaithersburg, MD 20899)

Estimates of ΔI are reported for amplitude changes in $F2$ of two-formant synthetic vowels and for amplitude changes in $F3$ for three-formant synthetic vowels. The vowels /æ, ɔ, i, and u/ were used. Four subjects were tested at an overall level of 70 dB SPL using both a same–different and a 3AFC procedure. The results suggest that ΔI for a "comparison" stimulus is dependent, in part, on the amplitude of $F2$ (or $F3$) relative to the amplitude of $F1$ in the "standard" stimulus. Thus, for example, when the dB level of $F3$ is low relative to the dB level in $F1$ (as in /u/), ΔI is comparatively large. When the amplitude of $F3$ is high compared to $F1$ (as in /æ/), ΔI is comparatively small.

D13. Duration effects revisited: Labeling of tone analogs to voicing contrasts. Richard E. Pastore, Crystle Morris, Robert Logan, and Jody K. Layer (Department of Psychology, SUNY University Center, Binghamton, NY 13901)

Our previous research described at the last Acoustical Society Meeting [Pastore *et al.*, J. Acoust. Soc. Am. Suppl. 1 **77**, S27 (1985)] investigated duration effects in the perception of tone analogs to the delayed $F1$ onset cue for voicelessness. Two groups of psychophysically experienced subjects exhibited statistically significant dependencies on stimulus duration, with longer stimuli resulting in shorter delayed onset boundaries. This pattern of results is opposite to those reported for voicing contrast boundaries as a function of target stimulus duration. The second group of subjects now has replicated this statistically significant reversed duration dependency for equivalent stimuli with added harmonic structure. Naive subjects with the identical task and stimuli exhibit a statistically significant dependency on duration consistent with published findings for synthetic speech, and thus opposite to that found for our experienced subjects. The results are discussed in terms of the role of perceptual learning in the perception of voicing cues. [Research supported, in part, by National Science Foundation grant 8302873.]

D14. Abstract withdrawn.

D15. Formant transitions as partial invariants in the identification of voiced stops. T. M. Nearey, S. E. Shammass, and M. L. Dow (Department of Linguistics, University of Alberta, Edmonton, Canada T6G 2H1)

The $F2$ trajectories for /b,d,g/ in CV syllables are often summarized by initial $F2$ frequency ($F2i$) and steady-state vowel $F2$ ($F2v$). Trajectories were measured for 330 Canadian English CV syllables (3 stops × 11 vowels × 10 speakers.) Plots for each stop (vowels pooled) indicated a strong linear relationship between $F2i$ and $F2v$. A regression line fitted to each plot represents an invariant relational property of the corresponding consonant, and $F2$ trajectories are not sufficient to uniquely specify the stops, since the lines for the three consonants intersect (indicating category overlap). However, the slopes and intercepts for the three consonants are distinct and thus represent "partly distinctive invariant properties" or "partial invariants." Similar patterns obtain for $F3$. Use of partial invariants of $F2/F3$ trajectories in a classification algorithm (based on distance from category lines) results in an identification rate over 70%. Extensions of the algorithm to include spectral shape information will be discussed, as will relationships to perceptual data.

D16. Some aspects of the auditory analysis of lateral consonants. Anthony Bladon and Peter Burleigh (Phonetics Laboratory, University of Oxford, 41 Wellington Square, Oxford, OX1 2JF, United Kingdom)

From its consistent presence in sweep-frequency measurements and in natural speech data, the spectral antiformant in the 2700-Hz region might be thought to be a candidate perceptual feature involved in distinguishing laterals from vowels. However, in isolated synthetic steady-state stimuli, the presence versus absence of an antiformant is barely detectable. This finding can be interepreted (consistent with other experiments using fricatives) as indicating that the auditory filter "smoothes over" the spectral notch. The width of the notch was exaggerated experimentally, as a probe for the width of the auditory filter in speech. A value in excess of 1 bark, and consistent with the assumption of a 3- to 3.5-bark-wide filter, was found. This property of auditory analysis seems to underlie a rather large range of speech effects. A second question which arises is whether the antiformant might be detected by temporal auditory mechanisms, such as enhanced onset/offset of discharge in the appropriate channel. But, when stimuli with and without the antiformant were embedded in an intervocalic context, hardly any evidence of perceptual ehhancement was found: Differences were again barely detectable. Instead, apparently, the contrast lateral versus vowel relies on grosser auditory characteristics such as transition duration, steady-state duration, and the amplitude envelope.

D17. Effects of speaking rate on the perception of syllable-initial stop consonants. Allard Jongman (Department of Linguistics, Box 1978, Brown University, Providence, RI 02912)

It has been shown that the effect of speaking rate on the perception of voice onset time (VOT) is strongest in the immediate vicinity of the target stop consonant: Listeners interpret a given VOT value in accordance with the duration of the CV syllable containing the stop consonant [A. Q. Summerfield, J. Exp. Psychol. Human Percept. Perform. **7**, 1074–1095 (1981); J. L. Miller *et al.*, J. Acoust. Soc. Am. Suppl. 1 **76**, S89 (1984)]. These studies used stimuli that are unlikely to occur in natural speech. The first study used simplified synthetic VOT continua; the second study used computer-edited natural-speech VOT continua, but in which syllable duration was kept constant, so that vowel duration decreased as VOT increased. The present study explored the effects of speaking rate on the perception of syllable-initial stop consonants in more natural circumstances. Natural-speech [bi–pi] continua were constructed, in which vowel rather than syllable duration was kept constant. Preliminary results indicate that the phoneme boundary is not, or only minimally, affected by changes in vowel duration.

D18. Shadowing fluent speech: The effects of mispronouncing word-initial consonants. Larry H. Small (Department of Communication Disorders, Bowling Green State University, Bowling Green, OH 43403-0233) and Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701)

A shadowing task was employed to investigate normal listeners' ability to detect alterations in word-initial consonants. The same 600-word prose passage, containing 14 two-syllable target words with syllable-initial stress, was employed in three separate experimental conditions so that the word-initial consonant was either altered by (1) a one distinctive feature change [Chomsky and Halle, *The Sound Pattern of English* (Harper and Row, New York, 1968)], (2) a three distinctive feature change, or (3) completely deleting the word-initial consonant. The target words were selected to be predictable from context and were examined for frequency of occurrence in English. There were 16 subjects in each condition. The subjects' shadowed responses were recorded on tape and analyzed to determine whether certain lexical or phonetic information was particularly salient in guiding the listener's shadowing behavior. Specifically, subject's responses were analyzed as to whether the part of speech, morpheme boundary, number of syllables, stressed vowel, and lexical stress were maintained when shadowing the mispronounced target words. Each of these items will be discussed as to their relative importance in lexical access while shadowing fluent speech.

D19. Subphonemic mismatches and perceiving by segments or by syllables. Keith Johnson (Department of Linguistics, Ohio State University, Columbus, OH 43210)

Reaction time measurements to tokens composed by means of categorial phonetic mismatches have indicated that information which is distributed throughout a syllable is used in making segment identification judgments [Whalen, Percept. Psychophys. 35, 49–64 (1984)]. This raises the possibility that speech perception proceeds from syllable perception "backwards" to segment identification. The mismatches created using tokens similar to those of Whalen's experiment 1 are not identical in their demands on the listener. Thus the mismatched fricative may be closer or further from the expected fricative depending on the degree of coarticulatory rounding. For instance, if the formant transitions lead the subject to expect a palatal fricative, the fricative noise of [us] will be closer to the expected fricative than that of [is]. It is proposed in this paper that the perception by syllables approach would predict that subjects would recover from the mismatch more quickly when the actual fricative noise is close to the expected fricative while the perception by segments would predict the opposite. The results of an experiment testing these hypotheses support the perception by segments approach. The interaction (in an ANOVA) between actual fricative and original vocalic context of the fricative was significant [$F(3,42) = 3.52, p < 0.05$] in the appropriate direction.

D20. Stream segregation effects in the perception of syllables. Laurie F. Garrison and James R. Sawusch (Department of Psychology, State University of New York at Buffalo, Buffalo, NY 14226)

Perceptual grouping processes and their role in the perception of speech syllables were investigated using repeated presentations of syllables. This method, often referred to as "streaming," has previously been used to study perceptual grouping processes for various nonspeech, acoustic stimuli. This experiment utilized a wide variety of synthetic consonant–vowel and consonant–consonant–vowel syllables. Some of these stimuli contained formant transitions while other stimuli consisted of adjacent noise (burst) and vocalic segments with no formant transitions. The effects of stimulus structure upon stream segregation and the implications of these results for selective adaptation studies will be discussed. [Work supported by NINCDS.]

D21. Intelligibility of speech in a noisy environment. Bruce M. Sanders and Thomas D. Rossing (Physics Department, Northern Illinois University, DeKalb, IL 60115)

Articulation indices for various modes of speaking masked by varying levels of white and pink noise have been measured. Pink noise is found to be a more effective masker than white noise at the same level. Shouted speech is more easily masked than normal speech presented at the same level. Very little difference is found between an actor's "stage voice" and normal speech. Enhancing speech in the 2- to 4-kHz range (region of the "singer's formant") also makes little difference in intelligibility in a noisy environment.

D22. Integral processing in the perception of phonemes within and across syllables. Gail R. Tomiak and James R. Sawusch (Department of Psychology, State University of New York at Buffalo, Buffalo, NY 14226)

Analogs of speech syllables were used in a speeded classification task with subjects instructed to treat the stimuli as speech or as nonspeech. Previous research using this paradigm with noise-tone analogs of monosyllables and disyllables has shown that single syllable stimuli show integral processing of adjacent phones (speech) but separable processing of adjacent pitch and amplitude information (nonspeech). Furthermore, this integrality effect is more pronounced within syllables than across syllable boundaries. We suggested that this pattern of results reflects a speech mode of processing in which phonetic information and knowledge of the consequences of coarticulation is largely represented within syllabic units. To assess the generality of these previous findings, new analogs of monosyllabic and disyllabic speech stimuli were constructed and tested in the speeded classification task. Results will be discussed in relation to the use of coarticulatory information in speech processing. [Work supported by NINCDS.]

D23. Contributions of the lexicon and sentential context to speech perception. Cynthia M. Connine (Indiana University, Bloomington, IN 47405 and Speech Research Laboratory, MIT, 36-511, Cambridge, MA 02139)

In experiment one, we investigated the influence of lexical status on speech perception first demonstrated by Ganong [J. Exp. Psychol.: Human Percept. Perform. 6, 110–125 (1980)]. Two voice onset time (VOT) series were presented to listeners for identification: DICE–TICE, the voiced endpoint forms a word, and DYPE–TYPE, the voiceless endpoint forms a word. Similar to Ganong, we found a longer VOT boundary value for the voiced compared to the voiceless lexical bias series. Reaction times for labeling responses showed an advantage for word compared to nonword responses for ambiguous stimuli (at the category boundary) but no word advantage for clear stimuli (at the continua endpoints). Two additional experiments investigated sentence context effects on phoneme identification in a word–word voicing continua (e.g., DENT–TENT). Each stimulus appeared as the final word in sentences pragmatically biased toward the voiced and voiceless endpoint. Voiced biased contexts resulted in longer VOT boundary values than voiceless biased contexts. Reaction times showed an advantage for pragmatically appropriate responses at the continua endpoints and the boundary. The lexical and sentence context effects will be discussed within modular and interactive theoretical frameworks. [Work supported by NIMH.]

D24. Perception of stress contrasts by the hearing impaired. Judith Rubin-Spitz, Nancy S. McGarr, and Karen Youdelman (Center for Research in Speech and Hearing Sciences, 33 West 42nd Street, Room 902, New York, NY 10036)

Waveform manipulation and LPC resynthesis techniques were used to develop test stimuli in which the $F0$, amplitude, and durational cues for first versus second syllable stress were systematically and independently controlled. Both normal and exaggerated stress contrasts were represented in the test corpus. Twenty severely to profoundly hearing-impaired students served as subjects for a perceptual experiment and were asked to indicate whether the stress was on the first or second syllable of each two-syllable test item (data were also collected on a group of normal hearing controls). Results revealed that the hearing-impaired listeners were able to correctly identify the stressed syllable only for stimuli in which the amplitude cue was available. That is, they scored below chance on stimuli in which the durational and/or $F0$ cues were present but where the amplitude cue had been neutralized. Exaggerating the amplitude cue for stress resulted in higher identification scores. Exaggerating $F0$ and/or durational cues did not improve performance. The implications of these results will be discussed. [Work supported by PHS grant No. NS17764-05 to the Graduate Center, C.U.N.Y.]

D25. Speech pattern audiometric assessment of hearing-impaired children. V. Hazan and A. J. Fourcin (Department of Phonetics, University College London, 4, Stephenson Way, London NW1 2HE, Great Britain)

A microprocessor-controlled speech pattern audiometer has been applied to study the long-term speech perceptual development of a group of hearing-impaired children. This technique makes use of high-quality synthetic speech to form minimal pairs of increasing acoustic complexity. The efficiency of the labeling tests is improved through the use of an interactive procedure. Individual children's development in the ability to process a range of increasingly complex contrasts has been assessed over a 3-year period. Development of perceptual ability is found to be similar to that of normally hearing children but delayed. The order of development is explainable in terms of the acoustic complexity of the stimuli. The primary use made of specific cues, such as the first formant in the labeling of the vowel contrast, and spectral rather than temporal cues in the labeling of the voicing contrast, is highlighted. The importance of intersubject differences in the development of perceptual abilities is also stressed.

D26. The effect of increased spectral contrast on vowel intelligibility for hearing-impaired listeners. M. F. Dorman and M. R. Leek (Hearing Research Laboratory, Community Service Building, Arizona State University, Tempe, AZ 85287)

At the previous meeting of the Society, we reported that hearing-impaired listeners required larger peak-to-valley differences in vowel spectra for correct identification than do normal-hearing listeners. This outcome suggests that errors in vowel identification might be reduced by increasing the contrast between spectral peaks and valleys of natural vowels. To test this hypothesis, we created three sets of four vowels: one set with normal peak-to-valley amplitudes, one set with valley amplitudes decreased by 5 dB, and one set with valley amplitudes decreased by 10 dB. Identification accuracy improved minimally with 5-dB enhancement but not at all with 10-dB enhancement. The failure to markedly improve identification accuracy suggests that the deficiency in vowel identification does not lie in locating the spectral peaks, but rather may be due to aberrant coding of frequency in the internal auditory representation.

D27. Temporal processing and speech perception in reading disabilities. Marjorie A. Reed (Psychology Department, Cleveland State University, Cleveland, OH 44115)

This study examined the processing of speech and nonspeech sounds by 23 second and third grade reading disabled children and 23 age- and sex-matched controls. Tallal's [Brain and Lang. **9**, 182–198 (1980)] finding that reading disabled children were found to be poor in discriminating 75-ms tones and in making temporal order judgments about the tones was replicated. The difficulty in process in brief stimuli was found to extend to stop consonant temporal order judgments but not to vowel stimuli. The stop consonant syllables were distinguished by brief formant transitions, while the vowels were differentiated by long, steady-state frequency differences. The present study also demonstrated that the deficit does not extend to the visual modality, nor does it reflect generally poor performance on all tasks. Reading disabled children were also impaired in discriminating words distinguished by initial stop consonants and showed less sharply defined boundaries between phonological categories on a place-of-articulation continuum, demonstrating that the deficit observed in the temporal order tasks influences the perception of speech in natural situations and the clarity of phonological representations.

D28. Children's discrimination of CV syllables differing in VOT: II. Lois L. Elliott (Audiology and Hearing Impairment, Northwestern University, Evanston, IL 60201)

Normal children aged 6–8 and 8–11 years and young adults were tested on a same–different task using a [ba]–[pa] continuum with VOT's that ranged from 0–35 ms. Data were analyzed in terms of d', using tables of Kaplan *et al.* [Behav. Res. Methods Instrum. **10**, 1978 (1978)]. The adult subjects' discrimination was better than the children's for pairs of CV's differing by 10 and 20 ms; this supported results previously obtained using a different test paradigm and different subjects (Elliott *et al.*, Child Dev. (in press)]. Calculated k values indicated that adults demonstrated greater bias than children. Results were interpreted as indicating true age differences in discriminating these stimuli. [Work supported, in part, by NINCDS and NIH.]

TUESDAY MORNING, 13 MAY 1986 RITZ ROOM, 8:30 TO 11:40 A.M.

Session E. Shock and Vibration I: Statistical Energy Analysis

Joseph M. Cuschieri, Chairman

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

Chairman's Introduction—8:30

Invited Papers

8:35

E1. Some aspects of the statistical energy analysis—SEA. G. Maidanik (David Taylor Naval Ship Research and Development Center, Bethesda, MD 20084-5000)

Recently SEA has been gaining recognition as a bonafide noise control tool. A few of its past critics are not only rescinding, but are joining in its advocacy. Naturally SEA has, since its inception and development in the early 1960s, been improved, extended, and even reformulated. The advantages and limitations of its use are now more clearly understood and defined. In this paper some recounting of SEA is presented, removal of a few early limitations reviewed, and a few items for further consideration are discussed.

9:00

E2. Power flow techniques in the study of machinery installations for the purpose of vibration control. R. G. White (Institute of Sound and Vibration Research, University of Southampton, Highfield, Southampton 509-5NH, England)

A typical machine installation consists of a source of vibration mounted on resilient isolators attached to a flexible substructure. There are usually also other vibration transmission paths, such a pipework, shafts, etc., which act as short-circuit elements across the isolators. It can thus be seen that there are a variety of mechanisms by which vibration is transmitted from the source to the point of interest on the substructure. The power flow approach to this type of problem is a basic concept which enables the relative importance of the various transmission paths to be critically assessed. Work has been carried out on the estimation of vibrational power flows between coupled systems, using approximate mobility methods, and this has led to the establishment of design rules for machinery seatings, together with simple formulas for estimating point mobilities of structures. Experimental techniques have also been developed. One development concerns measurement of vibrational power flow through isolators so that, for one machine mounted on a set of isolators or even an array of machines, the power inputs to the substructure at each isolator connecting point can be determined. The other, most recent development is the structural intensity meter which enables one- and two-dimensional power flows to be measured in beams and plates. Intensity maps can be plotted for plate-type structures, for example. The paper outlines the power flow approach to the machinery installation problem, with the objective of vibration control, and reviews the developments noted above.

9:25

E3. Energy accountancy concept. E. J. Richards (Center for Acoustics and Vibration, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The study of noise sources on an engineering level calls for a good working knowledge of machinery mechanics, structural vibration, energy absorption or damping, and fluid mechanics as well as acoustics. The acoustic output is effectively a remainder term associated with inefficiency in one or many of these processes. Due to uncertainties existing between various subjects, new mathematical techniques used to obtain exact calculations of cause and effect cannot be exploited. It is the contention of this paper that mathematical methods dealing with special simplified cases should be associated with a systems analysis which is strong in concept and universality rather than in detail. The energy accountancy system is intended to be such, not specifying the details of how each component is studied, but how they are linked together. It is based on the idea that noise occurs only when compressibility occurs in the flow near to a body or flow, and that in noisy machines this can be equated to the high-acceleration portions of the motion. It uses an energy balance approach which allows one to ignore the exact modal vibratory front except where necessary, as well as ignoring coupling and directional effects. It looks at a single machine impact and adds a repetition effect, because a large number of high-frequency modes arise from the sharp accelerations which can only be dealt with statistically. Thus while SEA can and is used in some of the terms in the "energy accountancy equation," the method is not dependent upon this.

9:50

E4. Power flow through dynamically coupled subsystems. Richard N. Brown (BBN Laboratories, Inc., 10 Moulton Street, Cambridge, MA 02238)

In practical applications of the statistical energy analysis (SEA) or other power flow techniques, the calculation of the power flow between subsystems is of fundamental concern. When the subsystems are connected by multimodal structures, for instance, vibration mounts, values for the coupling loss factors as found in the literature are not suitable. In this paper we present a general theoretical framework for calculating the coupling loss factor when the coupler is numerically modeled by classical dynamics, notably by means of a finite element analysis (FEA). After presenting some pertinent results and definitions from SEA, FEA, and structural dynamical theory, an expression for the "coupling" impedance matrix, which relates the forces and free velocities at nodes along the subsystem/coupler boundaries, is derived. This impedance is formed from the admittance matrix of the coupler, calculated using FEA, and input impedances for the systems, analytically calculated. From the coupling impedance, expressions for the coupling loss factors are derived for the cases where the SEA subsystems are plates or beams and include flexural, longitudinal, and torsional forms of energy transmission. Finally, numerical examples are given for the application of this technique to structures consisting of two beams or plates connected by a multimodal coupler.

10:15

E5. Applications of SEAM to the noise analysis and design problems. Richard G. DeJong (Cambridge Collaborative, Inc., 689 Concord Avenue, Cambridge, MA 02138)

A procedure for applying the principles of statistical energy analysis (SEA) to practical noise analysis and design problems has been developed and implemented in the computer program SEAM. Ths user models a system by specifying the geometric and material properties of subsystems. SEAM computes the necessary SEA parameters, such as modal density, impedance, and coupling loss factor, and then solves for the system response to a particular excitation. Examples are given of the application of SEAM to a variety of complex dynamic systems showing comparisons with measured results.

10:40

E6. Response of a thin-walled pipe section due to external excitation using energy accountancy I: Analytical study. J. M. Cuschieri, R. F. Schapley, II, and S. E. Dunn (Center for Acoustics and Vibration, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The energy accountancy method has been shown by Richards *et al.* to be a very practical and useful tool to estimate the noise energy radiated by a machine structure in broad frequency bands. The radiated noise energy was estimated to within ± 3 dB for structures with a high modal density. For structures with low modal density, the method was used to estimate the total radiated noise energy and the mean level of radiated noise. This method of use of the energy accountancy concept is very much similar to statistical energy analysis. However, unlike statistical energy methods which are mainly restricted either to high modal density structures, where one can use statistical averaging to model the behavior of the structure or to estimate the total level of radiated noise for low modal density structures, the energy accountancy method can be used to estimate the noise energy radiated by a structure in narrow frequency bands. In this paper, using an energy accountancy approach, an expression is developed for each of the different energy components based on a series solution of the response of the structure. Through this analysis it is shown that the energy accountancy approach can indeed be used to estimate the radiated noise energy from a structure in narrow frequency bands.

10:55

E7. Response of a thin-walled pipe section due to external excitation using energy accountancy II: Experimental study. R. F. Schapley, II, J. M. Cuschieri, and S. E. Dunn (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The response of a thin-walled pipe section excited by an external force can be obtained using the energy accountancy method as described in the first part of this series of two papers. In order to verify this approach, an experimental study was conducted on a 2.88-m-long, thin-walled steel pipe section containing two different fluids, air and water. The pipe section, 0.06 m in diameter and 0.003 m thick, was simply suspended by wires with no other outside connections. The pipe was excited externally by a calibrated hammer with the response of the pipe measured by an accelerometer attached to the outer surface of the pipe. In these experiments the internal fluid was stationary. This did not impose any limitations on the understanding obtained from the experiments since, for water systems, the flow is at a very low Mach number and therefore the effects of the flow are negligible. The results obtained show good agreement between the measured energy input and the measured dissipated and radiated energy components, the energy levels being within 1 dB of each other. This verified the application of the energy accountancy concept in describing the response of a thin-walled pipe section. Furthermore, the results obtained can be simplified into a diagnostic tool which can be used for noise control purposes.

11:10

E8. Energy flow and acoustic radiation for a fluid-loaded panel. P. L. Maillet, J. M. Cuschieri, and S. E. Dunn (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

The results of studying the energy flow in and the acoustic radiation from a fluid-loaded panel using the energy accountancy method are reported. Energy components are defined for the excitation, structural dissipation, and acoustic radiation in terms of excitation parameters, the structure, the medium, and the fluid-structural coupling factor. Using the energy components, a balance equation is written. The method is evaluated for a point excited flat plate with clamped edges. The plate is subjected to air loading on one side and water loading on the other. A comparison is made between the predicted energy balance terms using the accountancy approach and experimentally measured results. The energy balance equation provided good results between 50 Hz and 12 kHz.

11:25

E9. Prediction of helicopter cabin noise using statistical energy analysis (SEA). James A. Moore (Cambridge Collaborative, Inc., 689 Concord Avenue, Cambridge, MA 02138)

The noise environment in the cabin of small 6–13 passenger commercial helicopters is dominated by mesh tones generated within the gearbox. Helicopter airframes are efficiently designed to support mechanical loads which results in their also being efficient transmitters of vibratory energy. Structural components of the airframe are characterized by many resonant vibratory modes at higher frequencies in the audio range. Statistical energy analysis (SEA) has proved to be a very useful tool in the development of a model of airframe vibration transmission and coupling to the acoustic environment in the cabin. Under contract support from NASA, Langley, an SEA model of the Sikorsky Aircraft S-76 helicopter was developed and implemented using the Cambridge Collaborative's general purpose SEAM computer code. Characterizations of the structural connections between frame members, and frame members and adjacent skin panels, were developed in implementing the model. Additional coupling loss factors between panels, frames, and acoustics spaces are needed in building the SEA model. Measurements on actual airframes with shaker excitation during flight operating conditions were performed with good agreement between measured and predicted vibration and cabin noise levels.

Session F. Underwater Acoustics I: Ray Theoretic Methods in Acoustic Modeling

Michael Brown, Chairman
RSMAS—AMP, University of Miami, Miami, Florida 33144

Chairman's Introduction—8:30

Invited Papers

8:35

F1. Rays aren't what they used to be. L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic Institute of New York, Farmingdale, NY 11735)

Departing from the classical formulation of the propagation of light rays, modern ray theory has evolved by successive improvements and now provides one of the more effective methods for treating high-frequency wave propagation in general—in various environments and disciplines—in terms of field contributions localized around ray trajectories. By asymptotic analysis of *canonical problems* in the geometrical theory of diffraction (GTD), the category of incident, reflected, and refracted rays has been enlarged through inclusion of diffracted rays accounting for edge scattering, critical angle phenomena, and creeping waves in shadow regions on smooth convex surfaces. By asymptotic uniformization, failures of GTD in transition regions near shadow boundaries, caustics, or foci have been repaired; by collective treatment, multiple reflected ray fields in guiding regions have been converted into ray congruences that represent the normal modes; and from partial conversion of poorly converging ray clusters, there have emerged self-consistent and physically incisive hybrid ray-mode combinations. By extending the notion of a ray into complex coordinate space, it has been possible to incorporate evanescent fields and highly collimated beam fields within the ray format. Very recent studies have shown that the common foundation for all of these ray phenomena is a local plane-wave spectrum which is most severely shrunk around the ray paths when the ray fields are simple, but which must retain more "spectral flesh" in transitional and other critical domains. These concepts, which also extend to time-dependent propagation, are explored in the presentation, with emphasis on the motion of generalized ray fields as spectral objects. [Work supported by ONR.]

9:00

F2. Gaussian beam synthetic seismograms. Robert Nowack (Department of Geosciences, Purdue University, West Lafayette, IN 47907)

This presentation is a tutorial on the Gaussian beam method used for the asymptotic synthesis of seismic and acoustic wave fields in inhomogeneous media. The method is based on the superposition of beam solutions, each of which is an approximate solution of the wave equation along particular rays. Smoothness conditions on the medium are required for the approximate propagation of beam solutions. Within a smoothly varying medium, various choices of the beam parameters can be used. Specifying broad planar beams at the source would result in a plane-wave decomposition of the visible spectrum. The standard ray method would result by specifying narrow planar beams at the receiver. Another choice is to specify the minimum integral beam width along each ray. There are several advantages of the Gaussian beam method, including finite amplitudes at caustics, smoothing of endpoint errors, and the reduction of amplitude variability resulting from model parameterization. Several numerical examples will be given in 2-D and 3-D, and comparisons will be shown with other numerical methods.

9:25

F3. The modified Cagniard method and its application to the calculation of impulsive acoustic waves in a layered fluid. Adrianus T. de Hoop (Laboratory of Electromagnetic Research, Department of Electrical Engineering, Delft University of Technology, P. O. Box 5031, 2600 GA Delft, The Netherlands)

Through a specific combination of a Laplace transformation with respect to time and Fourier transformations with respect to the "horizontal" spatial coordinates (the modified Cagniard method), explicit space-time expressions are obtained for the acoustic pressure and the particle velocity of an acoustic wave generated by an impulsive source in a "vertically" layered fluid. The expressions have the form of a time convolution of the wave shape of the source strength and a properly defined acoustic space-time Green's function. The representation for the Green's function is either an explicit algebraic one (for two-dimensional wave motion), or a finite integral of an algebraic expression (for three-dimensional wave motion). The general aspects of the method are presented and examples are discussed.

9:50

F4. Maslov's method in seismology. Colin Thomson (Programs in Geosciences, University of Texas-Dallas, Richardson, TX 75080 and Queen's University, Kingston, Ontario, Canada)

Classical ray theory is an asymptotic form of a more general wave solution for smoothly inhomogeneous media, one which is uniformly asymptotic (i.e., valid at caustics) and has the WKBJ seismogram as a special case. This method involves "Fourier integral operators," which exploit well-known properties of wave fronts and rays. Physically interpreted, it accounts for rays that miss the receiver or, equivalently, it is a local sum of plane waves, looking like a finite Fourier transform. Phases and amplitudes for these "plane" waves are derived from adjacent ray travel times and amplitudes by Legendre transformation and canonical transformation, respectively. Time domain seismograms are obtained using the efficient algorithm of Chapman: Most work is in ray tracing, not waveform construction. Smooth boundaries are accounted for using plane-wave coefficients. This covers, e.g., the intersection of a caustic and reflector or the Fresnel transition into the shadow of a grazing ray. Here, R/T coefficient discontinuities at critical and grazing angles cause low-frequency errors, as do finite integration limits, but these are recoverable.

10:15

F5. A history of ray-theory development at Naval Ocean Systems Center. M. A. Pedersen (Computer Sciences Corporation, 4045 Hancock Street, San Diego, CA 92110) and D. F. Gordon (Naval Ocean Systems Center, San Diego, CA 92152-5000)

This paper presents examples of the evolution of ray theory of underwater acoustics at the Naval Ocean Systems Center and its predecessor laboratories since 1952. The motivations for this evolution were to bring ray-theory results into better agreement with experiments, to distinguish true acoustic properties of the ocean from artifacts resulting from the sound-speed model or from shortcomings in ray theory, and to develop simple controls for use in evaluating complicated propagation models. Examples of such controls are the five-parameter Epstein profile, profiles for which various ray pencils focus at a point, and closed-form solutions for profiles with range as well as depth dependence. The most significant topics presented are the development of continuous-gradient sound-speed profiles, the effect of Earth's curvature and of improvements in experimental sound speeds, the application of uniform asymptotics in the boundary layer about caustics, and the development of a method using complex parameters for the evaluation of shadow-zone fields. Various examples of ray-theory propagation losses are compared with experiment or with the results of normal-mode theory.

Contributed Papers

10:40

F6. Foundations of rigorous ray tracing. Edward R. Floyd (Arctic Submarine Laboratory, Naval Ocean Systems Center, San Diego, CA 92152-5000)

Rigorous ray tracing may be derived by inserting an ansatz into the Helmholtz equation to develop an alternate Hamilton–Jacobi equation for rigorous ray tracing expressed by

$$\left(\frac{\partial W}{\partial z}\right)^2 + C_m^{-2} - C^{-2} = \frac{-1}{2\omega^2}\,\frac{\partial^3 W/\partial z^3}{\partial W/\partial z} + \frac{3}{4\omega^2}\left(\frac{\partial^2 W/\partial z^2}{\partial W/\partial z}\right)^2,$$

where W is Hamilton's characteristic function (the generator of motion), z is depth, C is the sound velocity profile, C_m is the constant of motion (vertex velocity), and ω is the radial frequency. The left side of the above equation manifests the Hamilton–Jacobi equation for classical ray tracing. The terms on the right side of the above equation, which contain the common factor ω^{-2}, compensate for finite wavelength. This alternate Hamilton–Jacobi equation does not have any associated auxiliary equation and may be related to the original Hamilton–Jacobi equation for rigorous ray tracing [E. R. Floyd, J. Acoust. Soc. Am. 75, 803–808 (1984)]. Equations of motion for both ray paths and wave normals follow from the generator of motion W.

10:55

F7. A comparison of normal mode, WKBJ, and ray theory predictions with measured pulse arrival patterns at the 300-km range. Joseph J. Romm[a] and Guy Masters (Scripps Institution of Oceanography, A-025, La Jolla, CA 92093)

Precise measurements of the impulse response of the ocean sound channel were made during the 1983 reciprocal acoustic transmission experiment at the 300-km range with source and receiver near the sound channel axis. The transmitted pulse was centered at 400 Hz and had approximately 10-ms resolution. Using measured XBT data and the technique of inverse theory, a range-dependent sound-speed field had previously been computed such that the measurements were consistent with the prediction of range-dependent ray theory (Howe *et al.*, 1986). Predictions made for the range-averaged sound-speed profile using normal mode theory, the WKBJ approximation, and ray theory were found to be in excellent agreement (though a small discrepancy with the measurements remained, since range dependence was neglected). By allowing for range dependence through the use of adiabatic normal mode theory, this discrepancy was largely removed. We conclude that ray theory or WKBJ theory is an adequate prediction technique at 400 Hz in this geometry. [Work supported by NSF and ONR.] [a] Visiting from M.I.T.

11:10

F8. A Hamiltonian approach to horizontal ray tracing in adiabatic normal mode modeling. Richard Pitre and Robert F. Gragg (Naval Research Laboratory, Code 5160, Washington, DC 20375-5000)

Recently, Arnold and Felsen [J. Acoust. Soc. Am. Suppl. 1 78, S23 (1985)] and Kamel and Felsen [J. Acoust. Soc. Am. 73, 1120–1130 (1983)] have extended adiabatic mode theory to treat trapped-to-leaky mode transitions. These papers introduce a new spectral decomposition theory by identifying an adiabatic invariant for the discrete modes and then analytically continuing it to the whole complex horizontal wavenumber plane. The theory is formulated for waveguides composed of isovelocity layers with slowly varying thicknesses. Our work extends Felsen and Kamel's definition of the adiabatic invariant to arbitrary vertical sound-speed profiles by applying Milne's amplitude and phase representation of Sturm–Liouville solutions [Phys. Rev. 35, 865–867 (1930)]. This adiabatic invariant was employed to trace horizontal ray paths. In our approach, the horizontal ray paths are identified as the trajectories of

a two-dimensional dynamical system in which the Hamiltonian is the adiabatic invariant and the momentum is the horizontal wavenumber. By analytically continuing that Hamiltonian, ray paths can be defined for modes which pass through cutoff. A numerical model has been developed for horizontal ray tracing in waveguides with arbitrary vertical structure and slow but otherwise arbitrary horizontal variations. With this model the horizontal refraction of adiabatic modes for some waveguides of interest is examined.

11:25

F9. A coherent shallow water eigenray model. Arnold W. Novick (Mission Sciences Corporation, 6090 Jericho Turnpike, Commack, NY 11725)

An efficient range-independent coherent shallow water propagation loss model based on eigenrays is described. The model uses logic which guarantees finding all ray paths. Travel time data are also accurately computed. Predictions at 64, 256, and 1024 Hz for a complex shallow water environment (140-m water depth) are compared with measured data, and predictions using a detailed geoacoustic model (data and geoacoustic data reported by D. D. Ellis and D. M. F. Chapman, Defence Research Establishment Atlantic). Propagation loss predictions out to 85 km are in good agreement with measurements at 64 and 256 Hz using this relatively simple range-independent ray tracing model with a multilayer bottom. Surprisingly, the predictions made at 1024 Hz, where ray theory may be expected to be more accurate, are not quite as good. Comparison of model predictions with other measured and theoretical data is also presented. The results generally support the validity of coherent ray-based propagation modeling in shallow water areas over a wide frequency range.

11:40

F10. Adaptive acoustical tomography. William Mansfield Adams (Department of Geology and Geophysics, University of Hawaii, Honolulu, HI 96822)

Acoustical and seismological tomography differ significantly from tomography as practiced using energy which travels in straight lines, such as with x rays. The complication of refraction introduces a large number of new variables. This has the indirect result of reducing the variance for most of the variables. Usually overlooked, conveniently, is that the variances are no longer equal, i.e., the variance of the variances is not a constant. A novel approach to experimental design, adaptive in nature, is revealed. This permits, but does not require, the investigator to proceed adaptively in the taking of observations. Extreme canonical examples are illustrated.

Session G. Physical Acoustics II: Chaotic Phenomena

David T. Blackstock, Chairman
Applied Research Laboratories, University of Texas, P. O. Box 8029, Austin, Texas 78713-8029

Special Invited Lecture

1:15

G1. Chaotic phenomena in acoustic and elastic wave transmission lines. Francis C. Moon (Theoretical and Applied Mechanics, Cornell University, Ithaca, NY 14853)

A review of chaotic dynamics in nonlinear oscillators will be given. Evidence for chaotic phenomena in coupled nonlinear oscillators and wave propagation systems will be presented. In particular, large amplitude vibrations in flexible elastic and acoustic vibrations in a cylindrical cavity will be examined. In the latter problem, nonlinear boundary conditions at the end of the tube produce chaotic modulation of an acoustic carrier signal when excited by a deterministic harmonic signal. How higher-frequency acoustic waves can excite low-frequency chaotic vibrations of an elastic end plug of the tube will also be shown. New experimental methods for analyzing chaotic dynamics will be discussed, including Poincare maps and fractal dimensions. Speculation on the existence of chaotic waves in other wave systems will be posited. A demonstration of chaotic oscillations will also be presented.

TUESDAY AFTERNOON, 13 MAY 1986 RITZ ROOM, 2:25 TO 4:45 P.M.

Session H. Engineering Acoustics I and Noise II: Electroacoustic Noise Cancellation: Active and Passive

Harry B. Miller, Chairman
Code 3234, Naval Underwater Systems Center, New London, Connecticut 06320

Lawrence J. Oswald, Vice Chairman
Engineering Mechanics Department, GM Research Laboratories, Warren, Michigan 48090

Chairman's Introduction—2:25

Invited Papers

2:30

H1. Adaptive sound control—A tutorial review. John C. Burgess (Department of Mechanical Engineering, University of Hawaii, 2540 Dole Street, Honolulu, HI 96822)

Adaptive methods are being implemented increasingly to control the response of systems to meet performance objectives in the presence of unpredictable time-varying constraints. Typical open- or closed-loop control systems often fail under such conditions because their controllers (filters) have fixed coefficients. Adaptive controllers have time-variable coefficients determined on-line to minimize the difference at a system's output between desired and actual outputs. Adaptive methods are particularly appropriate for sound control. It is well known that active fixed-coefficient methods for sound cancellation fail with time as they become detuned by time-varying constraints such as changing medium temperature and motion. The purpose of this tutorial review is to provide some of the background of adaptive control with specific application to electroacoustic sound cancellation.

3:00

H2. A generalized adaptive control approach to electroacoustic noise cancellation. David C. Swanson and Jiri Tichy (The Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16804)

Electroacoustic noise cancellation is instigated using a general adaptive control system similar to those used in controlling industrial processes. An optional control signal in the form of a sampled electrical input to a

loudspeaker is calculated via least-squares state estimation so as to minimize the variance of the measured control system output representing the sampled electrical output of a microphone. The control system is modeled as an auxiliary input autoregressive moving-average (ARMAX) process where the characteristics of the noise as well as the transducers are included in the system identification process. The mathematical properties of ARMAX control systems have been well studied [G. Goodwin and K. Sin, *Adaptive Filtering, Prediction, and Control* (Prentice-Hall, Englewood Cliffs, NJ, 1984)] and the closed-loop system will be stable in practice so long as the acoustic propagation time delays are known and a lower bound on the model order can be assumed. Numerical simulations indicate a significant improvement in noise reduction at the microphone when the more complicated ARMAX model is used over ordinary least-squares. The algorithm can be expanded for multiple inputs and outputs permitting sound to be canceled at several microphone locations by one or more loudspeakers.

3:30

H3. Appraisal of an active noise control scheme in a 1-D waveguide. Richard J. Silcox (NASA Langley Research Center, M.S. 463, Hampton, VA 23665)

An active noise control scheme was implemented for broadband random noise in a 1-D reverberant environment. A frequency domain approach was used with an adaptive algorithm to provide noise control in a 50- to 400-Hz band using conventional microphones and loudspeakers. A 256-tap transversal filter provided the necessary narrow-band control for the system to define and maintain a 15- to 30-dB reduction in transmitted power. Measured system frequency response functions defining both performance and control functions are related to primary system response functions, i.e., propagation parameters, transducer characteristics, and termination conditions. Results indicate reduced noise control near resonance in this lightly damped system. The use of a simple omnidirectional source, even with highly directional sensors, yields strong resonances when highly reflective termination conditions exist. In addition, the presence of control source feedback weakens the discrete frequency nature of the controller frequency response. The effect of finite source impedance is also examined. Previous work has considered only constant volume velocity sources in the development of active control schemes. Analytical results show the impact on system control functions and experimental results indicate that source impedance effects are important near speaker resonance.

4:00

H4. An active noise reduction system for flying helmets and ear protectors. Graham M. Rood (Human Engineering Division, Flight Systems Department, Royal Aircraft Establishment, Farnborough, Hants, United Kingdom)

One of the major limitations of flying helmets or hearing protectors is their inability to provide adequate acoustic attenuation at low frequencies. In both fixed and rotary wing aircraft and in armored fighting vehicles, the high levels of low-frequency noise cannot be suitably reduced by the use of passive hearing protectors alone. The development of active noise reduction for use in such protectors allows much of this limitation to be removed. This paper describes the development and testing, both in the laboratory and during flight trials, of an active noise reduction system fitted to the current UK military force's flying helmet. The system was developed under a Ministry of Defence contract by the Institute of Sound and Vibration Research at Southampton University, and assessed by the RAE in a number of helicopters and during high-speed, low-level flight in a strike aircraft. The results from both the laboratory testing and from the flight trials are discussed, and the operational and medical advantages of using this type of active system summarized, both for military and civilian use.

Contributed Paper

4:30

H5. Prediction of optimal active noise controllers using boundary element methods. Robert J. Bernhard and Chris G. Mollo (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

Boundary element methods have been used as numerical approximations of boundary integral equations to predict the sound fields in complicated three-dimensional interior and exterior spaces. In this investigation the methods have been investigated for their capability to find and evaluate the optimal active noise controller for complex acoustic geometries. The boundary element methods are utilized to enforce the specified boundary conditions while solving for the optimal secondary source distribution necessary to minimize a given performance equation. For the current work the performance equation is a weighted sum of mean-square pressures at specified points. Thus the method can predict the optimal controller required to achieve local control using a single point, or to achieve a space averaged reduction by using a distribution of points. The formulation allows multiple secondary sources and either pressure, velocity, or normal specific acoustic impedance boundary conditions. The optimal secondary source's strengths and the insertion loss performance of several active controller configurations are illustrated.

Session I. Physical Acoustics III: Propagation: Atmospheric Layers

David T. Blackstock, Chairman
Applied Research Laboratories, University of Texas, P. O. Box 8029, Austin, Texas 78713

Invited Paper

2:30

I1. Propagation of normal acoustic modes in an atmospheric boundary layer. William E. Zorumski and William L. Willshire, Jr. (Acoustics Division, NASA Langley Research Center, Hampton, VA 23665)

The Obukhov quasipotential function for the acoustic field in a boundary layer of exponential profile is used to obtain a modal description of low-frequency sound propagation. As the wind speed approaches zero, the governing equations approach the Helmholtz equation with an impedance boundary condition. The solutions for the acoustic field with a boundary layer can be given as a continuous plane-wave spectrum with variable amplitudes given by generalized hypergeometric functions. An analysis with the hypergeometric functions gives one or more acoustic modes, depending on frequency. The acoustic modes propagate as cylindrical waves, with amplitude varying inversely with the square root of distance. An estimate of the wavenumber of the fundamental mode shows that its attenuation is proportional to the product of wind speed and boundary layer displacement thickness. The propagation theory is compared to data from a wind turbine at Medicine Bow, Wyoming. A microphone array was used to measure low-frequency sound at ground level at distances from 200 to 20 000 m from the turbine. Atmospheric temperature and wind speed profiles were measured, as was ground impedance, so that the theory may be compared without ambiguity to the data.

Contributed Papers

3:00

I2. Application of ray theory to downwind propagation of low-frequency noise in the atmosphere. James A. Hawkins, Jr., and David T. Blackstock (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

Willshire [J. Acoust. Soc. Am. Suppl. 1 77, S91–S92 (1985)] and NASA Tech. Mem. 86409 (April 1985) have reported measurements of downwind propagation for very low-frequency noise from a large wind turbine generator. The source height was 40 m, and ground measurements were taken out to 10 000 m. The noise was found to spread spherically near the source (out to 450 m) but cylindrically thereafter. We present ray theory calculations of the propagation loss for this experiment. Our computer program is adapted to the atmosphere from MEDUSA, a ray theory program developed for underwater sound [T. L. Foreman, Tech. Rep. ARL-TR-83-41, Applied Research Laboratories, The University of Texas at Austin, 1983, ADA-137202]. We assume a realistic ground impedance to calculate reflections at the ground and a logarithmic wind velocity profile to account for sound-speed variation. Results for 8 Hz show that channeling caused by downwind refraction is responsible for the cylindrical spreading observed downrange of the source. Near the source, spherical spreading occurs because channeling has not yet been established. The results are in quantitative agreement with Wilshire's data. [Work supported by NASA.]

3:15

I3. Estimating the velocity vector of fast moving sound source exploiting the retardation effect. Joachim Schiller (FGAN-Forschungsinstitut für Hochfrequenzphysik, Abteilung SuK, Neuenahrer-Strasse 20, D-5307 Wachtberg-Werthoven, West Germany)

The radiated noise of moving sound sources can be used to detect and localize these objects. Using microphone arrays with apertures comparing to the wavelength of the received signals, localization and velocity vector estimation will generally be done by means of triangulation. In this presentation it will be shown that, under certain conditions concerning the parameters of motion, the velocity vector itself (and not only the angular velocity) can be reconstructed using only the correlation measurements of *one* microphone array consisting of four microphones. The microphones of the array form three noncollinear bases with base lengths on the order of 0.5 m. The additional information needed results form the retardation effect. The retardation effect is a consequence of the finite signal velocity, and results in a mismatch between the indicated and the real target position if the target is moving. It becomes important at speeds of the sound source comparing to the signal propagation speed. On the basis of field measurements of low-flying aircraft, the reconstruction of the dynamic flight parameters (i.e., velocity vector, flight height, impact parameter) on the basis of the correlation measurements at one microphone array will be demonstrated and compared with simulation results.

3:30

I4. Turbulence effects on sound propagation from an elevated source. Richard Raspet, Michael T. Bobak, and Mark A. Johnson[a] (U.S. Army Construction Engineering Research Laboratory, P. O. Box 4005, Champaign, IL 61820-1305)

In a study of sound propagation from an elevated source, the turbulence effects appeared to be much smaller than was expected from Daigle's paper on turbulence effects on sound propagation with source and receiver close to the ground [J. Acoust. Soc. Am. 65, 45–48 (1979)]. In this paper, we describe the application of an atmospheric model for turbulence developed by Johnson and Raspet [J. Acoust. Soc. Am. Suppl. 1 77, S92 (1984)] to the refractive turbulence effects theory of Clifford and Lataitas [J. Acoust. Soc. Am. 73, 1545–1550 (1983)]. This theory predicts the reduction of turbulence effects observed in the noise study for elevated sources. The implications of this study for aircraft noise measurements will be briefly discussed. [a] Presently with Watkins-Johnson, 2525 N. First Street, San Jose, CA 95131-1097.

3:45

I5. Diffraction of sound over curved ground. G. A. Daigle and T. F. W. Embleton (Division of Physics, National Research Council, Ottawa, Canada K1A 0R6)

Theory for predicting the diffraction of sound (creeping waves) into shadow regions, resulting from propagation in either a stratified medium or above a convex curved surface, is well known and has been used mostly in underwater sound. This theory has also been extended more recently to the case of a stratified atmosphere. However, this theory generally has not been used to predict the energy diffracted over curved ground, such as rolling hills on berms. More commonly, the diffracted energy in these cases has been predicted by adapting theory valid for thick barriers or wedges. The creeping wave theory may be more appropriate in cases such as the diffraction of sound over berms, or of sound of very low frequency over rolling ground. Preliminary experiments have been performed over convex curved (radius of curvature < 100 m) ground of finite and infinite impedance in the frequency range between 250 and 8000 Hz using propagation distances up to 30 m. Controlled measurements are also planned indoors over a carefully constructed curve surface. The measurements are compared with various aspects of theory.

4:00

I6. Effects of humidity on the characteristic impedance in air. George S. K. Wong (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

Recent investigations into the effects of humidity and temperature on the specific heat ratio γ in air have led to the prediction of the above effects on sound speed [G. S. K. Wong and T. F. W. Embleton, J. Acoust. Soc. Am. **76**, 555–559 (1984); **77**, 402–407 (1985); **77**, 1710–1712 (1985)]. With an approach which is similar to the above prediction, one is able to ascertain the variation of the characteristic impedance ρc in air with humidity and temperature. In standard atmospheric air, the numerical value of ρc decreases with the increase in humidity and temperature. An approximate equation will be presented for the calculation of ρc over the range of relative humidity from 0 to 1.0, temperature from 0° to 30 °C, and at 1 atm.

4:15

I7. Theoretical predictions of turbulent boundary layer pressure fluctuations transmitted into a viscoelastic-fluid composite layer. Sung H. Ko (Naval Underwater Systems Center, New London, CT 06320)

The objective of this paper is to develop a model for evaluating the turbulent boundary layer pressure fluctuations transmitted into a viscoelastic-fluid composite layer. The theoretical model considered here is a plane layer of viscoelastic-fluid composite structure backed by a rigid plane surface. The other side of the composite layer is exposed to turbulent flow. The transmitted flow noise received by a rectangular hydrophone embedded in the composite layer was calculated for the turbulent boundary layer wall pressure spectrum proposed by Corcos. The transmitted flow noise was characterized by the frequency spectral density expressed in decibels. The main results presented in this paper are turbulent boundary layer noise reductions, which are given relative to the noise levels evaluated in the absence of the composite layer. Effects of the viscoelastic layer thickness, the fluid layer thickness, properties of the viscoelastic layer, and flow conditions on the noise reduction are discussed.

4:30

I8. Application of the parabolic wave equation to atmospheric sound propagation over a locally reacting ground surface. Michael J. White and Kenneth E. Gilbert (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677 and Institute for Technology Development, 3825 Ridgewood Road, Jackson, MS 39211)

Although the parabolic wave equation (PE) is widely used in underwater acoustics, it has seen only limited use in atmospheric sound propagation. This paper assesses the usefulness of the PE for outdoor sound propagation over a flat, locally reacting ground surface. The investigation addresses three questions: (1) Can the PE accurately treat the boundary condition at a locally reacting ground surface? (2) What starting field is needed for the PE to accurately describe the sound field near the source? (3) Does the PE have sufficient phase accuracy to predict the field far from the source? In order to answer these questions, we compare PE calculations to benchmark calculations from a fast field program (FFP) and to measure data. The PE calculations presented are for a nonturbulent atmosphere. The extension of the calculations to account for turbulence is discussed.

4:45

I9. Resonances in the impedance curves for outdoor ground covers. James M. Sabatier and Henry E. Bass (Physical Acoustics Research Laboratory, The University of Mississippi, University, MS 38677)

Measurements of the outdoor ground surface impedance for frequencies below 200 Hz are sparse. Recent experimental measurements at low frequencies show "glitches" or resonances in the measured impedance values [for example, G. A. Daigle and M. R. Stinson, J. Acoust. Soc. Am. Suppl. 1 **78**, S86 (1985)]. Using a modified version of the Biot–Stoll model for wave propagation in poroelastic media (J. Acoust. Soc. Am., submitted) the impedance of these outdoor ground surfaces is predicted. The glitches are theoretically predicted and explained as p- and s-wave interferences within the weathered porous layer of the ground.

Session J. Physiological Acoustics II and Psychological Acoustics II: Binaural Hearing in Man and Animals

Richard M. Stern, Chairman

Department of Electrical and Computer Engineering and Biomedical Engineering Program, Carnegie–Mellon University, Pittsburgh, Pennsylvania 15213

Contributed Papers

2:30

J1. Performance on interaural time discrimination: Direction does make a difference. Sheila V. Stager (School of Human Communication Disorders, Dalhousie University, 5599 Fenwick Street, Halifax, Nova Scotia, Canada B3H 1R2) and Ted. L. Langford (Callier Centre for Communication Disorders, University of Texas at Dallas, Dallas, TX 75235)

The question of differences in subjects' performances on interaural time discrimination depending on which ear received the slight delay was explored. Twenty subjects with normal hearing sensitivity and no previous training on interaural time discrimination participated in the study. A "same/different" procedure was used to measure the discriminability of low-frequency noise bursts with interaural phase differences of 20 and 12 deg favoring first one ear, and then the other. From the two resulting psychometric functions, the interaural differences (in degrees) which produced a 75% correct level of performance were determined. Subjects could be classified according to their performances. Some subjects performed differently depending on ear favored. Some subjects performed equally well regardless of ear favored. Some subjects did not reach the criterion of 75% correct with the introduction of a phase difference of 20 deg regardless of ear favored. These results are considered in light of previous studies of interaural time discrimination, and of time- versus intensity-sensitive subjects (McFadden *et al.*, 1973). [Work supported in part by NIH grant #NS16396.]

2:45

J2. Binaural fusion, apparent motion, and the precedence effect. David R. Perrott (Psychoacoustics Laboratory, California State University, Los Angeles, CA 90032) and Thomas Z. Strybel (Department of Psychology, University of Arizona, Tucson, AZ 85721)

Wideband noise pulses, 50 ms in duration, were presented from pairs of speakers. Across sessions, the angular separation of the sources was varied between 6 and 160°. Within sessions, interstimulus onset intervals (ISOI) varied between 0.1 to 500 ms. Eight listeners were required to perform two concurrent discrimination tasks. The first was an objective procedure in which they were required to report the temporal order in which the sources were activated. The second task required them to assign the event to one of five descriptive categories: single stationary image; multiple stationary images; single moving image; an interrupted but moving image; or two successive images. Discrimination of the temporal order of speaker activation was only moderately affected by the angular separation of the sources (thresholds ranged between 12–20 ms). These results are well in line with other experiments in which temporal order judgments are required. However, when subjects report "apparent motion," a relatively rare event with small ISOI's, judgments of temporal order are remarkably accurate (exceeding 80% correct) with all but the shortest ISOI's employed. The implications of these results will be discussed. [Work supported by NSF.]

3:00

J3. The accuracy of head saccades under monaural and binaural listening conditions. J. Tucker and David R. Perrott (Psychoacoustics Laboratory, California State University, Los Angeles, CA 90032)

Literature examining head motion and auditory function has typically focused upon whether changes in position facilitate auditory spatial performance. The present experiment addresses the reverse of this question by attempting to quantify the spatial characteristics of the head saccade itself, in the context of various localization tasks. Horizontal head position was measured every 16 ms as both binaural and monaural listeners turned to face a sound source located along an arc of + or − 60 deg azimuth. The signals were 0.5-, 2.0-, and 3.7-kHz tones presented at 40 dB SPL. Both single pulse and pulse train conditions were employed. Resolution was best with speakers near the subjects median plane for binaural conditions, and on the side opposite the occluded ear in monaural conditions. Precision of the final head position could be predicted by the frequency used. The current results follow known auditory spatial functions and support the 81-year-old hypothesis that while binaural localization is superior to monaural, given the opportunity to move one's head in the presence of an ongoing acoustic signal, monaural listeners may still be quite accurate. [Work supported by NIH.]

3:15

J4. Localization of pure tones in the front–back and vertical dimension under monaural and binaural conditions. Alan D. Musicant (Department of Neurophysiology, University of Wisconsin Medical School, Madison, WI 53706), J. Tucker, and David R. Perrott (Psychoacoustic Laboratory, California State University, Los Angeles, CA 90032)

Butler and Flannery (1980) proposed the spatial referent map theory. The present experiment was designed to test this theory under extreme conditions. Presentation of pure tones in the monaural condition should stimulate the spatial referent map directly. Here 500-ms pure tone pulses were used. Frequencies ranged from 0.5 to 9.5 kHz. Speakers, circumaural earphones, or insert earphones were utilized. The method of paired comparison with a four-alterative, forced-choice response task was employed. Subjects were requested to respond on both front–back and vertical dimensions. Without any possibility of interaural difference cues, tones did appear to come from different spatial locations. Despite a difference in the monaural condition the free field performances supported Butler's theory. In all conditions, responses on the vertical dimension showed a monotonic relationship between frequency and phenomenal vertical position. The data on the front–back dimension were relatively stable across subjects in the monaural condition, however, the between-subject variability was too great to make any kind of definitive statement.

3:30

J5. Binaural detection and discrimination: Normal listeners. Carrin Passaro, Janet Koehnke, and H. Steven Colburn (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139 and Biomedical Engineering Department, 110 Cummington Street, Boston University, Boston, MA 02215)

In conjunction with a study of binaural performance in impaired listeners, the performance of normal listeners in several binaural tasks were measured. Specifically, just-noticeable differences (jnd's) were measured in interaural correlation (IC), interaural time delay (ITD), and interaural intensity difference (IID) for one-third octave noise bands centered at 500 Hz and 4000 Hz. In addition, we measured NoSπ detection thresholds at the same two frequencies for a tone centered in a third-octave band of masking noise. Measurements were made at several reference values of ITD and IID, including -300, 0, and $+300\,\mu$s and -12, 0, and $+12$ dB. jnd's and thresholds were estimated using a relatively crude adaptive method so that a complete set of measurements could be obtained quickly for each subject. Results are discussed in terms of the dependence on reference interaural conditions and in terms of the ability to predict performance on each task from performance on the other tasks. So far, the only consistent effect of changing interaural reference parameters is larger jnd's in IID with reinforcing values of ITD and IID. Also, IC discrimination performance is directly predictable from the detection threshold and vice versa. The ability to predict both of these tasks from ITD and IID discrimination data is evaluated for various theoretical models. [Work supported by NIH.]

3:45

J6. Binaural detection and discrimination: Impaired listeners. Janet Koehnke and H. Steven Colburn (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139 and Biomedical Engineering Department, 110 Cummington Street, Boston University, Boston, MA 02215)

The experimental tasks, techniques, and stimuli for this study are the same as those described for normal listeners (see preceding abstract), except for the choice of levels at each ear. A set of reference interaural intensity differences (IID's) were chosen based on IID's corresponding to equal SPL, equal sensation level (20 dB SL), equal loudness (ABLB measurements), and a centered perception. Performance in the four binaural tasks was measured at reference interaural conditions that include all combinations of these IID's and interaural time differences of -300, 0, and $+300\,\mu$s. Subjects in these experiments have varying degrees and configurations of hearing loss, and thus far binaural performance shows no clear relation to these audiometric measures. Data from illustrative subjects are discussed in terms of the dependence of performance on the reference interaural condition and in terms of the ability to predict results in one task from performance in the other tasks. The consistency of performance and predictions for the relation between NoSπ detection and interaural correlation discrimination is observed in the impaired listeners as well as in the normal listeners. [Work supported by NIH.]

4:00

J7. Detectability of tonal signals with changing interaural phase differences in the presence of diotic or interaurally uncorrelated noise. D. Wesley Grantham (Bill Wilkerson Hearing & Speech Center, 1114 19th Avenue South, Nashville, TN 37212)

Detectability of binaurally presented 400- and 800-Hz tonal signals was investigated in an adaptive, two-interval, forced-choice experiment. A continuous 3150-Hz low-pass noise masker was employed, which was presented either diotically (No) or interaurally uncorrelated (NU) at an overall level of 70 dB SPL. Signal duration was either 100 or 1000 ms. In part 1 of the experiment thresholds were determined for several values of signal interaural phase difference between 0° (So) and 180° (Sπ) [e.g., Jeffress, Blodgett, and Deatherage, J. Acoust. Soc. Am. **24**, 523–527 (1952)]. In part 2 the interaural phase difference varied dynamically during the signal presentation (slightly different frequencies were presented to the two ears). The range of interaural phase variations was selected to yield the same varying interaural temporal differences that would be produced if real auditory targets traversed various arcs in the horizontal plane. The data will be discussed in terms of previous data related to dynamic binaural processing, with particular regard to the potential effect of an auditory object's movement on its detectability. [Supported by NIH.]

4:15

J8. Perception of modulations in pitch and lateralization. Laural Beecher and Richard M. Stern (Department of Electrical and Computer Engineering and Bioengineering Program, Carnegie–Mellon University, Pittsburgh, PA 15213)

The ability to perceive low-frequency sinusoidal modulations of monaurally and dichotically created pitch was compared to the perception of modulations of the subjective lateral position of a binaural image. The stimuli used in the dichotic pitch experiments had low-pass spectra that were flat between 0 and 2000 Hz, and they were created by modulating the time delay of a multiple-phase-shift filter [F. A. Bilsen, J. Acoust. Soc. Am. **59**, 467–468 (1976)]. Stimuli with similar spectra and sinusoidally modulated interaural time delays (ITD's) were used for the latralization experiments [D. W. Grantham and F. L. Wightman, J. Acoust. Soc. Am. **63**, 511–523 (1978)]. Also examined was the perception of monaural frequency-modulated pure tones, and monaural low-pass stimuli with power spectra similar in shape to the "internal spectra" of the dichotic pitch stimuli. Subjects discriminated between sinusoidally modulated and unmodulated stimuli using two-cue, two-alternative 4IFC paradigms, and we determined the threshold frequency deviation or ITD as a function of modulation frequency. Results were similar in form for all experiments: threshold frequency deviation or ITD was constant for modulation frequencies up to a particular "corner" frequency, and then increased as a power function of modulation frequency up to at least 32 Hz. This corner frequency was 4 Hz or less for the binaural lateralization and dichotic pitch experiments, and approximately 10 Hz for the two monaural pitch experiments. [Work supported by NIH.]

4:30

J9. The acoustic role of the noseleaf in a bat emitting frequency-modulated signals. David J. Hartley and Roderick A. Suthers (School of Medicine and Department of Biology, Indiana University, Bloomington, IN 47405)

Carollia perspicillata is a New World frugivorous bat which emits low-intensity, broadband, frequency-modulated sonar pulses through the nostrils, which are embedded in a noseleaf structure. The emission pattern of this bat is of interest because the relationship between the nostril spacing and emitted wavelength should vary during the pulse, potentially causing complex interference patterns in the horizontal dimension. Sound pressures were measured around the bat using a moveable microphone and were referenced to a stationary microphone positioned directly in front of the animal. It was found that the emission pattern differed markedly from those of bats previously studied in that there were prominent sidelobes in the horizontal dimension. The pattern suggests an effective nostril spacing of around 0.75 λ at the frequency of maximal pulse energy (between 90 and 100 kHz). Interference between the nostrils was confirmed by blocking one nostril which eliminated the sidelobes. Displacement and manipulation of the dorsal lancet of the noseleaf showed that this structure serves to direct the sound in the vertical dimension. [Work supported by NSF.]

4:45

J10. What are fish listening to?—A possible answer. Peter H. Rogers (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Although all fish hear, relatively few species are known to vocalize. The biological relevance of hearing in fish is thus somewhat of an enigma. In this paper, we suggest that what fish may be listening to is ambient noise scattered from the swim bladders of nearby fish. This is in some ways analogous to the role of the visual system in most animals, where the relevant stimulus is ambient light scattered by objects rather than light emitted by luminous objects. Although the scattered noise signal is much weaker than the unscattered ambient noise signal, it was demonstrated that a relatively simple processing scheme could greatly enchance the S/N and enable a fish to detect and unambiguously localize a nearby fish from the scattered noise. The processing scheme, which relies on the nearfield properties of the scattered signal, is shown to be consistent with the known capabilities of the fish auditory system and may well explain many of them. [Work supported by ONR Code 420.]

Session K. Psychological Acoustics III and Speech Communication II: Speech Perception in Normal and Impaired Listeners

Murray F. Spiegel, Chairman
Bell Communications Research, 435 South Street, Morristown, New Jersey 07960

Contributed Papers

1:00

K1. Simultaneous and nonsimultaneous masking within natural speech. Murray F. Spiegel (Bell Communications Research, 435 South Street, MRE 2E-252, Morristown, NJ 07960)

Auditory masking functions are well known only for steady-state tones and noises, not for complex sounds such as speech. To investigate the contribution of simultaneous and nonsimultaneous masking in natural speech, thresholds were obtained for 15-ms probe tones placed in the closure portion of naturally spoken VCV utterances. The utterances contained a long closure (stop/d/) or a short closure (flap/P/). Two additional conditions were tested: flap closure was replaced by a short portion of stop closure, and the surrounding vowels were digitally attenuated. Critical-band filtering and spread of masking characterize the simultaneous-masking results. Some conditions reveal significant amounts of nonsimultaneous masking. These effects are less straightforward than the effects of simultaneous masking, primarily due to the time-varying nature of the natural speech maskers. Results of this line of research may help refine perceptually weighted filters used in speech coders to reduce the perceived level of quantization noise [cf. M. R. Schroeder, B. S. Atal, and J. L. Hall, J. Acoust. Soc. Am. **66**, 1647–1652 (1979)].

1:15

K2. Band importance functions for certain consonant features. Vasanta Duggirala, Gerald A. Studebaker (Department of Audiology and Speech Pathology, Memphis State University, 807 Jefferson Avenue, Memphis, TN 38105), Chaslav V. Pavlovic (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242), and Robert L. Sherbecoe (Department of Audiology and Speech Pathology, Memphis State University, 807 Jefferson Avenue, Memphis, TN 38105)

Miller and Nicely (1955) demonstrated that there are differences in the most important frequency regions for the recognition of different linguistic features. The present study was designed to determine the relative importance of each one-third octave band between 178 and 8912 Hz for each of several consonant features. The test instrument was the diagnostic rhyme test (DRT) of Voiers, Cohen, and Mickunas (1965). The analytical methods of French and Steinberg (1947) were used to derive band-importance functions for the DRT as a whole and for each of the features, voicing, nasality, sustention, sibilation, graveness, and compactness. The one-third octave band importance weights for the DRT and the various features were found to be quite different from each other and from those reported earlier for nonsense syllables (ANSI S.3.5-1969; Kryter, 1962). Curves describing the importance per Hertz and the cumulative band importance will be presented. [Work supported by NIH grant # NS 15996.]

1:30

K3. The effect of four types of filtering on the intelligibility and quality of speech-in-noise for hearing-impaired listeners. Arlene C. Neuman and Teresa Schwander (Center for Research in Speech and Hearing Sciences, Graduate School, CUNY, 33 West 42 Street, New York, NY 10036)

A common complaint among hearing aid users is of difficulty in using amplification in noisy situations. In this study, subjective judgments of quality and intelligibility of speech-in-noise, as well as speech recognition scores, were obtained from sensorineural listeners through four different filters (1) flat frequency response, (2) a frequency response which places the rms level of the speech at the listener's comfort level, (3) a high-pass filter in combination with filter #2, and (4) a filter designed to keep the noise spectrum parallel to the listener's thresholds for one-third octave bands of noise. Preliminary results indicate that for individual listeners the four filters were ranked similarly, whether the criterion used was intelligibility or quality. Speech recognition performance was in agreement with subjective judgments of intelligibility. [Work supported by NIHR Grant No. G008302511.]

1:45

K4. Correlations between auditory capabilities and measures of phoneme perception. B. Espinoza-Varas, C. S. Watson, and D. A. Geddes (Department of Speech and Hearing, Indiana University, Bloomington, IN 47405)

Performance of 32 normal-hearing listeners on the test of basic auditory capabilities [Watson *et al.*, J. Acoust. Soc. Am. Suppl. 1 71, S73 (1982)] was correlated with each of the following measures of phoneme perception (obtained with the CUNY Nonsense Syllable Test): (1) *hit probability* for the overall test, and for each individual phoneme; (2) $P(C)_{max}$ for identification of each individual phoneme; and (3) $P(C)_{max}$ for discrimination of each possible phoneme pair within a set of phonemes. Relative to the total number of correlations that can be calculated, the percentage of *significant* correlations ($p<0.05$, $n=32$) were 0% for overall hit probability, 10% for phoneme-specific hit probability, 17% for identification $P(C)_{max}$, and 10% for pairwise discrimination $P(C)_{max}$. These results support the hypothesis [Espinoza-Varas and Watson, J. Acoust. Soc. Am. Suppl. 1 **78**, S47 (1985)] that auditory capabilities may be correlated more strongly with the ability to process specific phonemes than with overall measures of speech processing (e.g., whole-test speech discrimination scores). It was also found that a number of phonemes yielding either near-chance or near-perfect performance contribute little or nothing to the correlations or, presumably, to the information provided by the overall speech score. [Work supported by NIH and AFOSR.]

2:00

K5. Speech reception in noise by hearing-impaired listeners. P. M. Zurek and L. A. Delhorne (Research Laboratory of Electronics, Room 36-736, Massachusetts Institute of Technology, Cambridge, MA 02139)

The goal of this study was to determine the extent to which the difficulty experienced by impaired listeners in understanding noisy speech may be explained merely on the basis of elevated detection thresholds. Twenty impaired ears of 14 subjects, spanning a variety of audiometric configurations with average hearing losses to 75 dB, were tested for reception of consonants in a speech-spectrum noise. Speech level, noise level, and frequency-gain characteristic were varied to generate a range of listening conditions. Results for impaired listeners are compared to those of normal-hearing listeners tested under the same conditions with extra noise added to approximate the impaired listener's thresholds. Although there are a few exceptions, the conclusion based on this sample of moderate-to-severe hearing loss is that, when compared to normals listening

under similar conditions of threshold shift (or, more generally, similar values of articulation index), hearing-impaired listeners exhibit little or no handicap in speech reception. [Work supported by NIH.]

2:15

K6. Stop consonant recognition and audibility in normal and hearing-loss subjects. Christopher W. Turner and Michael P. Robb (Communication Sciences and Disorders Program, Syracuse University, Syracuse, NY 13244-2280)

This study examined the effect of spectral cue audibility on the discrimination of CV stop consonants in normal-hearing and hearing-impaired adults. Six synthetic speech tokens, each differing only in the initial 40-ms consonant portion, were presented to subjects in randomized lists. Performance-intensity functions and relative information transmitted were calculated from the results of a six-alternative, closed-set response task. In both normal-hearing and hearing-impaired subjects, recognition performance as a function of level differed among the six consonants. The first 40 ms of each CV were analyzed via FFT using an exponential window. The resulting spectral array was passed through a sliding-filter model of the auditory system to account for the proportional bandwidth filtering properties of the ear. This allowed the spectral data to be displayed in comparison to a subject's pure-tone thresholds. Differences in the amount of predicted audible spectral speech cues resulted in observable differences in the recognition performance of individual consonants.

2:30

K7. Auditory presentation of fundamental frequency as an aid to lipreading. Laurie Hanin, Arthur Boothroyd, and Terry Hnath (Graduate School, City University of New York, 33 West 42 Street, New York, NY 10036)

The probability of work recognition, in sentence context, was measured in eight normal subjects under two conditions: lipreading alone, and lipreading supplemented by fundamental voice frequency ($F0$). Test materials were 48 sets of sentences, each set containing 12 sentences varying in length from 3 through 14 words. Sentences were video recorded by a female talker, one audiochannel containing the acoustic speech waveform, and the other channel containing the output of an electroglottograph. This second channel, low-pass filtered at 360 Hz, provided the $F0$ supplementation. Subjects were informed of the topic of each sentence, but were not given feedback on performance. Some learning was observed during the initial stages of testing, mostly in the supplemented condition. After the scores stabilized, mean scores were 32% words correct for lipreading alone and 76% words correct for supplemented lipreading. Application of probability theory shows that the addition of $F0$ is equivalent to multiplying, by 3.7, the sources of statistically independent information in the lipreading stimulus. These data illustrate the high potential for auditory, tactile, visual, or electrocochlear presentation of $F0$ as an aid to lipreading in the postlingually deaf. [Research funded by NIH grant # 17764.]

2:45

K8. Learning disabled children's ability to discriminate time-compressed phonemes in sentential stimuli. Marie M. Watson and Michael P. Rastatter (Department of Communication Disorders, Bowling Green State University, Bowling Green, OH 43403)

This study measured the ability of a group of 8-and 12-year-old learning disabled children to discriminate phonemic contrasts in sentential stimuli presented at a 50% time compressed rate. These responses were compared statistically to similar data gathered from a group of 6-, 8-, and 10-year-old normal children and adults. Results were interpreted as suggesting that the learning disabled children exhibit auditory processing capacities reminiscent of an earlier level of operation, but also manifest marked differences in their feature and frequency processing abilities when compared to normally achieving children.

TUESDAY AFTERNOON, 13 MAY 1986 — EAST BALLROOM, 1:00 TO 5:05 P.M.

Session L. Speech Communication III: Text-To-Speech and Other Lecture Presentations

Astrid Schmidt-Nielsen, Chairman
Naval Research Laboratory, Code 7520, Washington, DC 20375

Chairman's Introduction—1:00

Contributed Papers

1:05

L1. History of text-to-speech conversion for English. Dennis Klatt (Room 36-523, Massachusetts Institute of Technology, Cambridge, MA 02139)

Text-to-speech conversion is now a practical technology with a history of research and development spanning at least 50 years. With the hope that valuable insight and perspective can be gained by examination of the evolution of a science, a literature search to trace the history of text-to-phoneme and phoneme-to-speech algorithms has begun. The paper will present a preliminary status report of this attempt to identify important milestones in the evolution of systems capable of generating intelligible English sentences from text, and trace the scientific lineage of selected commercial systems. A second potential benefit of relating technology to prior basic science is in identifying missing pieces in the puzzle as to why synthetic output is lacking in naturalness. Some conjectures concerning possible causes will be followed by suggestions for future research relevant to this domain. [Work supported in part by an NIH grant.]

1:17

L2. Preference judgments comparing different synthetic voices. John S. Logan and David B. Pisoni (Department of Psychology, Indiana University, Bloomington, IN 47405)

Speech from three text-to-speech systems was compared on the basis of listeners' preferences. Subjects heard a sentence produced by one system followed by the same sentence generated by another system. The task was to indicate which of the two voices they preferred and then furnish a

confidence rating for the decision. Each subject was presented with 40 pairs of sentences. Ten subjects participated in each of the three pairwise comparison conditions: (1) DECtalk compared with MITalk, (2) DECtalk compared with the Prose-2000, and (3) MITalk compared with the Prose-2000. The overall relationship among preference, response time required for this decision, and the confidence rating was examined. Results indicated a direct relation between preference and intelligibility whereas the remaining measures did not appear to be systematically related to preference. These subjective judgments will be discussed in terms of the relationships among preference, naturalness, and intelligibility in the perceptual evaluation of natural and synthetic speech. [Work supported by AFOSR and NIH.]

1:29

L3. Comprehension of natural and synthetic speech using a sentence verification task. Laura M. Manous, David B. Pisoni, Michael J. Dedina, and Howard C. Nusbaum (Department of Psychology, Indiana University, Bloomington, IN 47405)

A verification task was used to study sentence comprehension using two natural voices and five different synthetic voices generated by automatic text-to-speech conversion. Subjects listened to true and false, three- and six-word sentences produced by one of these voices. Sentence verification accuracy and speed, and sentence transcription accuracy were measured. A significant effect of voice type was obtained for true and false sentences for all three measures. In addition, for false sentences, subjects were less accurate in transcribing six-word sentences and they were slower in verifying these sentences. Furthermore, there were significant interactions of voice type with sentence length for all three dependent measures. Verification speed revealed a clustering of the seven voices into three basic categories corresponding to: (1) natural speech, (2) high-quality synthetic speech, and (3) moderate- to low-quality synthetic speech. Results from a second sentence verification task used to investigate effects of sentence predictability will also be discussed. [Work supported by AFOSR and NIH.]

1:41

L4. Segmental intelligibility of synthetic speech produced by eight text-to-speech systems. Beth G. Greene and John S. Logan (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Several studies measuring the segmental intelligibility of eight text-to-speech systems and a natural speech control using the modified rhyme test (MRT) have been conducted in the Speech Research Laboratory. Results indicated that the voices tested could be grouped into four categories: natural speech, high-quality synthetic speech, moderate-quality synthetic speech, and low-quality synthetic speech. The overall performance of the best system, DECtalk-Paul, was equivalent to natural speech only in terms of performance on initial consonants. The findings from these laboratory studies will be discussed in terms of recent work investigating the perception of synthetic speech under more severe conditions. Additional results obtained using a variation of the standard MRT will be presented. Suggestions for future research on improving the quality of synthetic speech will be discussed. [Work supported, in part, by NIH and AFOSR.]

1:53

L5. Diagnostic evaluation of a synthesizer's acoustic inventory. C. E. Wright, M. J. Altom, and J. P. Olive (Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

In speech synthesis, utterances are created by concatenating elements from an acoustic inventory made up of phonemes, dyads, syllables, or other convenient units. For further development of the AT&T text-to-speech system [J. P. Olive and M. Y. Liberman, J. Acoust. Soc. Am. Suppl. 1 **78**, S6 (1985)] we need a fast, simple method to identify problems with elements in the acoustic iventory. Phonetic transcription is one method that has been used in the past for this type of synthetic-speech evaluation. However, transcription experiments are difficult to run and require specially trained listeners. Other methods that have been used, e.g., the diagnostic rhyme test, are better suited to provide *comparative* information about several synthesizers than the *diagnostic* information needed for this evaluation. To fill this gap, we have developed a procedure that we call *verification*. It is an interactive, computer-based procedure that can be used with listeners who have little training. In this talk we describe the procedure and its early application.

2:05

L6. German text-to-phoneme software drives any speech synthesizer. W. Kulas and J. Blauert (Lehrstuhl für Allgemeine Elektrotechnik und Akustik, Ruhr-University Bochum, P. O. Box 102148, 4630 Bochum, Federal Republic of Germany)

The SYRUB system is a software system that converts any German text into phonemes. In addition to its high-quality conversion on word level, SYRUB automatically generates a German sentence intonation and controls its speech timing to obtain partial isochrony for more naturalness. Its universal synthesizer interface enables it to drive any speech synthesizer, thus allowing for an integration of the phoneme representations of seven European languages. Furthermore, special attention has been paid to the design of the system's user interface. The system distinguishes between trained and untrained users, and, subsequently supplies them with different sets of error and help messages, either acoustically or via screen. We will present the nonlanguage-specific aspects of our system: the user interface, the universal synthesizer interface, and the generation of speech rhythm.

2:17

L7. Evaluation of a synthesized Spanish accent. Deborah M. Rekart (Department of Speech-Language-Hearing: Sciences and Disorders, University of Kansas, Lawrence, KS 66045), Raymond G. Daniloff, Paul R. Hoffman, and C. J. Miller (Department of Speech Communication, Theater, and Communication Disorders, Louisiana State University, Baton Rouge, LA 70803)

English speakers' perception of a synthetic sentence with a moderate Spanish accent was reported on previously [J. Acoust. Soc. Am. Suppl. 1 **77**, S9 (1985)]. It was found that an increased number of cues caused significantly higher ratings, and formant frequency perturbation of full vowels was the strongest cue signaling accentedness. The present research investigated perception of newly synthesized strong and moderate accents, manipulating $F0$, VOT of syllable-initial voiceless stops, duration of sentence-medial stressed vowels, and formant frequency for full and reduced vowels. For each condition, the accented sentences were paired with the standard sentence in four randomizations. Forty-two English speakers rated how different each accented sentence was from the standard sentence and indicated confidence in their judgments. It was found that the synthetic English sentence was reliably rated for cue modifications indicative of a moderate Spanish accent; an increase in number of cues resulted in perception of increased accentedness and higher confidence ratings for both accents; $F0$ and formant frequency perturbation in full vowels were the most prominent cues signaling moderate Spanish accent, and their presence resulted in higher confidence ratings.

2:29

L8. Intelligibility of synthetic CVC stimuli over the telephone. Bathsheba J. Malsheen, James T. Wright, Melanie Yue (Speech Plus, Incorporated, 461 North Bernardo Avenue, Mountain View, CA 94043), and Margot Peet (Department of Linguistics, University of California, Berkeley, CA 94720)

Intelligibility tests of initial and final English consonants were made over a simulated long-distance telephone line for two leading text-to-speech converters descended from MITalk. CVC stimuli were presented to subjects in open-response listening tests to determine how vulnerable the intelligibility of synthetic speech would be to the effects of telephone bandwidth limitations. Subjects were also tested under nontelephone conditions for comparison. It was hypothesized that telephone bandpassing would produce a systematic breakdown in consonant intelligibility, and that certain phonemes, e.g., alveolar fricatives and alveolar stops before front vowels, would suffer the greatest loss due to their reliance on cues normally present at the higher frequencies. Our results show that overall intelligibility of high-quality synthetic speech is significantly reduced over the telephone. As expected, alveolar fricatives produced a large number of errors. Alveolar stops, on the other hand, remained robust and generated few perceptual errors. Velar stops, contrary to expectation, produced a large number of place and manner confusions. Intelligibility scores for initial and final consonants will be presented, and grouped by manner class.

2:41

L9. Synthesizing British English intonation using a nondownstep model. Briony J. Williams and Peter R. Alderson (IBM UK Scientific Centre, St. Clement Street, Winchester, Hants., SO23 9DR, England)

A method will be described for synthesizing British English intonation patterns from an initial representation in terms of tonetic stress marks. The resultant synthesized patterns compare favorably with those of the resynthesized originals. The data comprises fluently spoken whole sentences of natural speech, rather than unrealistically short utterances produced under artificial conditions. The tonetic stress mark system used is a modification of those used by O'Connor and Arnold (1961) and Crystal (1969). It is being used for large-scale prosodic transcription of a corpus of spoken English. Rules have been formulated for converting the tonetic stress marks to "target values" on a scale of one to ten, as in Pierrehumbert (1981). A declining topline and level baseline are then superimposed, yielding a frequency value for each syllable. The resultant $F0$ contour is compared with the original, for objective evaluation. The close match obtained suggests that tonetic stress marks are a valid starting point for intonation synthesis from annotated text. Since these units also carry functional weight, they may well be more satisfactory than an abstract tonal representation, for the purposes of speech synthesis from text.

2:53

L10. An investigation of mixed-source excitation models for linear predictive vocoders. Elizabeth A. Effer[a] and Stephen A. Zahorian (Department of Electrical Engineering, Old Dominion University, Norfolk, VA 23508)

Two methods were investigated for modeling the excitation for linear predictive vocoders as a mixture of pulses (periodic component) and noise (nonperiodic component). In one method, linear filters were used to separate the periodic and nonperiodic components of the residual signal. In another method, the two components were separated by locating the areas of peak energy in the residual and labeling the portion of the residual within each pitch period peak as the periodic component and the remainder of the signal as the nonperiodic component. Thus, in the first method, a frequency domain separation of the periodic and nonperiodic component was assumed whereas, in the second method, a time-domain separation of the two components was assumed. The time-domain method appeared to be more feasible. Speech was synthesized using a number of parameters to represent the two components of the excitation. Although some improvements in speech quality could be obtained by using very detailed representations of the pulses, less-detailed representations resulted in poorer quality speech than that obtained using a standard impulse excitation. [a] Currently at AT&T Bell Laboratories, Whippany, NJ.

3:05–3:29

Break

3:29

L11. On spontaneous speech and fluently spoken text: Production differences and perceptual distinctions. Robert E. Remez (Department of Psychology, Barnard College, 3009 Broadway, New York, NY 10027), Philip E. Rubin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06106), and Lynne C. Nygaard (Department of Psychology, Barnard College, 3009 Broadway, New York, NY 10027)

Our initial study [J. Acoust. Soc. Soc. Am. Suppl. 1 77, S38 (1985)] of spontaneously produced speech and fluently read speech found that these two kinds of utterance were distinguishable by untrained listeners, and may represent different modes of speech production. The perceptual differentiation of these types of utterance, which we tested using well-formed sentences free from gross metrical dysfluencies, seemed not to depend on a single acoustic emblem of spontaneity or reading, neither the presence or absence of pitch downdrift, nor average vocal pitch, nor average duration. The simplest acoustic correlate of perceptual success in identifying instances of spontaneous production, and of misidentifying read speech as spontaneous, was the variance of the phonatory frequency. Our present study again employs a perceptual test of the differences between spontaneous and read speech, using partial sentences. Listeners identified a test set composed either of the initial portions of sentence-size utterances or of the final portions. Partial sentences appear to offer information inferior to complete sentences about this distinction in most cases, though final portions are better sources than initial portions. The results are discussed with reference to speech production. [Research supported by NINCDS and NICHHD.]

3:41

L12. Assimilatory versus nonassimilatory neutralization processes in Catalan. Jan Charles-Luce (Department of Linguistics, Indiana University, Bloomington, IN 47405)

In general, phonologists claim that neutralization results in the acoustic-phonetic obliteration of an underlying phonemic contrast in favor of one the members of the contrast. For Catalan, two possible processes are involved in the neutralization of underlying voicing in word-final stops: (1) word-final devoicing, and (2) regressive voice assimilation. In the latter case, word-final stops are realized as either voiced or voiceless depending upon the voicing of the following phonetic segment. However, in the former case, word-final stops are always realized as voiceless in those environments that do not overlap with regressive voice assimilation. In the present investigation, native Catalan speakers produced monosyllabic minimal pairs differing in the underlying voicing of the final stop in two nonassimilatory environments and in two assimilatory environments. Vowel duration preceding the final stop, voicing during closure, and closure duration of the final stop were measured as acoustic correlates of voicing. The results bear on the question of whether complete neutralization may best be defined as an assimilatory or a nonassimilatory process in speech production. [Supported by NIH Training Grant NS-07134.]

3:53

L13. On the struggle of underlying vowels for a voice in surface phonetic structure—Evidence from Serbo-Croatian. Midhat Ridjanovic (Slavic Department, Ohio State University, Columbus, OH 43210)

An outstanding feature of the colloquial speech of Sarajevo in relation to standard Serbo-Croatian is the deletion of the high vowels [i] and [u] in post-accentual syllables. This gives rise to frequent "homonymy" of forms clearly distinguished in the standard language. The present work was undertaken to test the hypothesis that this homonymy is not perfect, i.e., that there could be phonetic remnants of the deleted vowels in adjacent segments. An experiment is being conducted in which (a) tape recordings of purportedly homonymous triplets of forms, two with deleted [i] and [u], respectively, and one involving no deletion, are presented to speakers of both the Sarajevo and the standard dialects of Serbo-Croatian for perceptual judgments, and (b) spectrographic analysis is carried out to discover possible mutual acoustic differences. Preliminary results have suggested that, in a majority of cases, the forms are preceived as if they contained the appropriate vowel [i] or [u] and that residual features of the deleted vowel may appear in the spectrograms of *both* contiguous segments.

4:05

L14. Distinctive length in initial consonants: Pattani Malay. Arthur S. Abramson (Department of Linguistics, University of Connecticut, U-145, Storrs, CT 06268 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The relative duration of the closure or constriction is the most salient physical manifestation of phonemically distinctive consonant length. The contrast is rare in the word-initial, and thus potentially utterance-initial position. In this context, perception would seem to depend upon the audibility of closure excitation. This is plausible for nasals, laterals, and fricatives. The closures of voiced stops, however, may have only low-amplitude excitation, while voiceless stops have none. Pattani Malay, a language of southern Thailand, was investigated to learn how robust the feature is in production and perception. Acoustic measurements of closure durations showed a reliable difference between the "short" and "long" consonants. Listening tests yielded good differentiation of the two length classes for isolated words, with a lesser effect for voiceless stops. Experiments with incrementally lengthened short closures and shortened long closures confirmed the sufficiency of duration as a cue. For the voiceless stops, these experiments could be run only for intervocalic position. Further work will seek other acoustic consequences of duration that might serve as cues. [Work supported by NICHD.]

4:17

L15. Cross-language switching in Dutch–English bilinguals. W. Eefting and James Emil Flege (Department of Biocommunication, University of Alabama at Birmingham, University Station, Birmingham, AL 35298)

A major issue in L2 research is whether bilinguals develop separate or merged phonetic systems. This study tested whether native speakers of Dutch, a language in which /p,t,k/ are realized with short-lag VOT values, would label the members of a VOT continuum ranging from /da/ to /ta/ differently when listening in a Dutch as opposed to English set. High-, mid-, and low-proficiency subgroups (ten each) were formed from a group of 50 Dutch university students on the basis of (a) global accent judgments by native British English listeners, (b) self-rating of ability to pronounce English, and (c) the VOT measured in their production of English /t/. A language specific set was established by having the subject produce and process Dutch or English speech prior to, or during, each half of the experiment. The mean location of the phoneme boundaries was only slightly longer in the English (35.6 ms) than Dutch set (33.5 ms). The effect of the language set was significant, but not the effect of proficiency or set x proficiency, indicating that the extent to which L2 learners adopt different, language-specific criteria in making phonetic judgments does not depend on language proficiency. However, the Dutch subjects showed a considerably larger shift in producing /t/ in English (60.3 ms) versus Dutch (22.8). English VOT was positively correlated with the global accent ratings, and subjects judged to have the least authentic English accent failed to distinguish /t/ in English and Dutch. [Work supported by NIH grant NS20963.]

4:29

L16. Perception of coarticulatory variation in Shona. Sharon Y. Manuel (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695 and Yale University, New Haven, CT 06511)

In Shona, a Bantu language, anticipatory vowel-to-vowel coarticulation is very strong in some VCV sequences, such as /api/. Two experiments were conducted to test whether native listeners are sensitive to these coarticulatory effects. The stimuli were versions of /api/, /apa/, /upi/, /upa/ in which the first vowel had been cross spliced from original /_pi/, /_pa/, or /_pu/ environments. In experiment 1, the task was to press a button whenever the second vowel was /i/. Subjects' responses were fastest when the first vowel carried the appropriate coarticulatory information (i.e., when it derived from the original /_pi/ environment). In experiment 2, subjects responded whenever the first vowel was /a/; they were slowest when the /a/ came from original /api/. Thus coarticulatory shifts in vowel quality facilitate the processing of later occurring information while, at the same time, they make the vowel itself less prototypical. These results, together with similar ones obtained by others for English, illustrate listeners' perceptual sensitivity to subphonemic phonetic variation. [Work supported by NICHD.]

TUESDAY PM

4:41

L17. Are $F0$ differences after stops accidental or deliberate? John Kingston (Department of Linguistics, Unversity of Texas, Austin, TX 78712-1196)

Differences in $F0$ after stops are generally thought to be unavoidable perturbations of the rate of vocal fold vibration brought about by changes in the tension of the folds during the stop [Hombert, Ohala, and Ewan, Language 55, 37–58 (1979)]. For example, $F0$ is depressed after voiced stops because the folds are slackened during the closure, but elevated after voiceless stops because the folds are tensed. Recent work has shown, however, that all the mechanisms proposed to account for differences in $F0$ after stops are seriously flawed [Kingston, MS thesis (1985)] and, furthermore, that stops of the same phonetic type, e.g., voiceless unaspirated, can elevate $F0$ in one language but depress it in another [Caisse, MS thesis (1982)]. The $F0$ differences may therefore be introduced deliberately, to enhance the phonological contrast, rather than arising as accidental byproducts of other articulations. This possibility was tested with data from Tamil, a language which contrasts long and short stops where the long stops are predictably voiceless and the short ones are predictably voiced. Since voicing is allophonic rather than distinctive in this language, $F0$ differences need not appear after stops if they are deliberate. On the other hand, if the $F0$ differences are simply an accidental byproduct of having the folds vibrating or not, then they should appear regardless of the phonological role of vocal fold vibration. Measurements of $F0$ after stops in Tamil show no significant difference at the onset of vowels following the two kinds of stops, supporting the contention that $F0$ differences are deliberate in other languages.

4:53

L18. English word and sentence stress patterns as spoken by non-native speakers. Joann Fokes (School of Hearing and Speech Sciences, Ohio University, Athens, OH 45701) and Zinny S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701)

Non-native speakers from six different language backgrounds and native American English speakers produced tokens of three-syllable words, such as *confession* and four syllable words, such as *confirmation* in isolation and in sentences. The acoustic–phonetic correlates of stress—amplitude, fundamental frequency, and duration—were measured for the first two syllables of each word. Both groups of speakers were highly variable. The Americans consistently produced stressed syllables with longer durations than unstressed, but varied considerably in their use of fundamental frequency and amplitude. Syllables of words in sentences were shorter than the words in citation form. In contrast, non-native speakers' fundamental frequency and amplitude variability was even greater, and they did not differentiate between stressed and unstressed syllables by means of duration. That is, their stressed syllables were too short and the unstressed too long. Lengthening of unstressed syllables in sentence context was a characteristic pattern of these non-native speakers.

Session M. Underwater Acoustics II: Ray Methods (continued) and Parabolic Equation Methods

Adrianus T. de Hoop, Chairman
Delft University of Technology, Postbus 5031, 2600 GA Delft, The Netherlands

Chairman's Introduction—2:30

Contributed Papers

2:35

M1. Acoustic propagation phenomena in time domain. R. H. Mellen (PSI Marine Sciences, New London, CT 06320)

Despite the intuitive appeal of ray theory, the associated time-domain properties remain largely neglected. Rays retain memory of "catastrophic" encounters along the path. A well-known example is polarity reversal at a pressure-release surface. Similarly, spherical waves resolve into an incoming and an outgoing impulse of reversed polarity, the waveform at the focus being the time derivative. Distortion of explosion waves, caused by refractive focusing, was first observed experimentally 25 years ago. For a cylindrical focus, phases advance $\pi/2$ and the outgoing impulse is Hilbert transformed while, on the caustic, the waveform is the Heaviside fractional derivative. Reflection at a fluid–fluid interface, for grazing angles less than critical, can also be expresed by Hilbert pairs $\delta(t)$ and $-1/\pi t$. For an elastic solid, the transmitted signal resolves into a shear and a compressional impulse. At the critical angle of total reflection, the two impulses merge to become partially evanescent. Comprehensive mathematical methods of treating all such phenomena have only recently begun to appear.

2:50

M2. Time series simulation in shallow water using eigenrays with beam displacement. Evan K. Westwood (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029) and C. T. Tindle (Physics Department, University of Auckland, Auckland, New Zealand)

A ray model for time series simulation in a multilayered, stratified environment has been extended to include the effects of beam displacement. Beam displacement is important in shallow water and occurs when the phase shift of the plane-wave reflection coefficient at an interface is a function of angle of incidence. The ray paths become frequency dependent, but the dependence is smooth so that a small number of frequency components (three or six) is sufficient to accurately estimate the transfer function of an eigenray with beam displacement. The caustics and shadow zones caused by beam displacement are also included. Simulated time series are compared with time series measured in a tank experiment having shallow water over a sand bottom. The significant ray paths have up to 14 bottom interactions, and agreement of the waveforms is good. The usual normal mode features of shallow water propagation are accurately reproduced. [Work supported by Independent Research and Development Program at Applied Research Laboratories, The University of Texas at Austin.]

3:05

M3. A new way to compute underwater acoustic bottom loss: Concept and application to borehole data. Craig A. Fisher and Frederick K. Duennebier (Hawaii Institute of Geophysics, University of Hawaii, Honolulu, HI 96822)

Bottom reflection loss cannot be computed with traditional methods using data from a borehole receiver. A grazing angle-dependent loss, unique to the borehole case, is incurred as a ray passes through the overlying sediments and basalts. To compute bottom loss from these data this grazing angle-dependent loss must be removed. Another angle-dependent loss, image interference, can also be present when the source is near an interface. We gave a bottom loss computation method which implicitly removes angle-dependent losses. In addition, our method is "self-calibrating" in both grazing angle and frequency. This technique uses multiple bottom-interacting rays with the same grazing angle to obtain redundant bottom loss estimates. The algorithm is applied to a data set recorded with a borehole receiver in the deep ocean at DSDP site 581C. For grazing angles from 20 to 60 deg and frequencies from 9 to 40 Hz we computed bottom reflection loss estimates of 1 to 12 dB, ± 2 dB. [Work supported by ONR.]

3:20

M4. Modeling underwater acoustic communications at moderate ranges. J. K. Thompson, K. Naghshineh, and K. B. Gates (4541 Briarcliff Trail, Copley, OH 44321)

This paper describes the analytical development and performance of a computer model of underwater sound transmission. This model is intended to simulate underwater acoustic communications over short to moderate ranges between subsurface transducers and buoys. Because of the high frequencies used in communication, the ray tracing method is employed in this simulation. The computations employed in this model are thoroughly described and compared to previous research. New developments of this study include a finite receiver dimension to account for the spreading of the acoustic energy in rays and an "eigenray" routine to minimize computation time and expense. Comparisons of calculations by this model and measured data were performed to determine the model's accuracy. The maximum predictive error of 5 dB would be acceptable in most communication applications. In no instance were extreme variations between the measured and calculated results encountered. [Work supported by NOAA Data Buoy Center of NSTL Station, Mississippi.]

3:35

M5. Shape of the wavefront which produces a transverse cusp diffraction catastrophe. Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

Diffraction patterns characteristic of a transverse cusp are known to be observable in light scattered from drops [P. L. Marston, Opt. Lett. **10**, 588–590 (1985)] or reflected from curved surfaces. It is to be expected that high-frequency sound reflected from curved surfaces or refracted by inhomogeneities may also produce transverse cusps but to facilitate their description one must know the shape of the outgoing wave which propagates to produce such a cusp. A wave whose amplitude in the xy plane is $\exp[ik(a_1x^2 + a_2y^2x + a_3y^2 - ct)]$ is considered and the two-dimensional diffraction integral which results from the Fresnel approximation of the Green's function is calculated. This diffraction integral reduces to one proportional to the Pearcey function $P(X,Y)$ or to $P^*(X,Y)$ depending on the sign of $a_1 + (2z)^{-1}$, where is the distance from the xy plane. The real parameters X,Y in the one-dimensional integral P depend on the a_j, z, k, and the transverse coordinates in the observation plane. This transformation locates the cusp where geometric optics fails. The shape of the wave differs from that which produces a longitudinal cusp present in certain convergence zones. [Work supported by ONR.]

3:50

M6. Improved paraxial methods for modeling underwater acoustic propagation. Thomas E. Bordley (Code 5123, Naval Research Laboratory, Washington, DC 20375-5000)

Current paraxial techniques for modeling wave propagation in the ocean perform poorly when bottom interaction is important due to their inability to handle wide-angle propagation. Such methods solve a simplified form of the acoustic wave equation, obtained by splitting the field into transmitted and reflected components, and then neglecting the reflected field. The concern of this work is to develop paraxial methods less restricted in angle, yet retaining the numerical stability and efficiency which have made this class of solutions so popular. The approach taken is to seek a split, in some sense, most consistent with the assumption that the modeled field closely approximates the "true" field. The measure of consistency adopted is the closeness of the split to the split implied by the field under the assumption that the parabolic and elliptic equations agree. An iterative algorithm is developed realizing this scheme which at each step, uses the current estimate of the field to generate a more consistent split for the next step. The new split is that which necessarily existed if the current estimate of the field were exact. The effectiveness of this approach is examined in the context of modeling sound propagation in the Arctic marginal ice zone.

4:05

M7. Notes on wide angle PE. L. Neil Frazer and Mrinal K. Sen (Hawaii Institute of Geophysics, 2525 Correa Road, Honolulu, HI 96822)

In many applications the parabolic equation method would be ideal were it not for the limitation to narrow angles. Recently, Fishman and McCoy [J. Math. Phys. **25**, 285–296 (1984)] used the theory of pseudodifferential operators to derive an integral equation for H, the square root of the operator $B=\omega^2/c^2(z)+\rho(z)\partial_z[\rho^{-1}(z)\partial_z]$, thereby obtaining an exact, arbitrary-angle PE theory. Numerical construction of the operator $H=\sqrt{B}$ remains an area of great interest because the integral equation for H is not easily solved. We show that the operator H can be numerically constructed to any desired accuracy by well-known methods of computational linear algebra. The key to this construction is that, whenever the sound-speed profile is real, the operator B is self-adjoint with respect to the inner product $\langle,\rangle$ where $\langle\phi,\psi\rangle=\int dz\,\rho^{-1}(z)\,\bar{\phi}(z)\psi(z)$. This means that the matrix of B can be diagonalized, hence, $H=\sqrt{B}$ and $\exp(i\Delta xH)$ are straighforward to compute. [Work supported by ONR.]

4:20

M8. Parabolic decomposition of the separable Helmholtz equation. David H. Wood (Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

The solution of the separable Helmholtz equation is represented as the convolution of solutions of two related (formally) parabolic partial differential equations. *Separable* means that the square of the index of refraction is a function of depth plus a function of range and that the depth of the ocean is assumed to be constant. This representation decomposes the separable Helmholtz equation into two parabolic partial differential equations. One equation is the usual Fock–Tappert parabolic equation for sound propagation. The solutions of the second parabolic equation may be regarded as candidate kernels of special integral transformations, or *transmutations*. All of these equations have many solutions, of course, so we present initial and/or boundary conditions that will determine solutions of the parabolic equations so that their convolution will give the Green's function for the Helmholtz equation. Examples of such transmutation representations will be presented.

4:35

M9. A range-dependent normal mode model with full mode coupling. E. Richard Robinson and David H. Wood (Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320; Department of Mathematics, University of Rhode Island, Kingston, RI 02881)

A preliminary version of a range-dependent transmission loss model is presented. It assumes that the ocean is piecewise constant in range. Full mode matching at the interfaces is used to compute the acoustic field. Our approach is similar to that used by Evans and Gilbert [Comput. Math. Appl. **11**, 795–805 (1985)] in that we also expand the desired normal modes as a weighted sum of depth-dependent basis functions. As is well known, this "Galerkin method" leads to a linear algebraic problem for the unknown weights. However, our approach is different because we insist on using other basis functions that lead to *structured* algebraic problems, where fast techniques are available. The future development of a fast version of this model will exploit this special algebraic structure. Implementation within the generic sonar model is used because this allows for considerable flexibility in updating developments and the choice of supporting submodels.

4:50

M10. Eigenray calculations in a uniformly stratified mutilayer medium. J. I. Arvelo (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20902-5000)

This paper presents the equations to be solved in order to calculate the travel time of an eigenray in a uniformly stratified refractive medium. The data required are the sound-speed profile and the relative position of the source and receiver. It has been proved that in a bilinear profile there exists up to three direct path eigenrays which can be calculated in a closed form by solving a cubic equation. However, in an N-layered media, it is not possible to predict the number of direct path eigenrays, nor calculate them, in a closed form. Ray tracing iteratively could be used to calculate these eigenrays. This eigenray method usually takes one to five iterations (less than 3 s) to obtain an accuracy in the horizontal range of one microfoot, and an uncertainty in the travel time of less than one microsecond, in a media of forty layers. Calculations were done on an HP9845B, 16 bit, 8 MHz, with interpretive BASIC language. The method described here is particularly useful in calculating the travel time difference of a system of one source (receiver) and a distribution of receivers (sources). A rapid and accurate travel time calculation is now possible with this method.

Session N. Architectural Acoustics: Discovering the World of Sound (Audiovisual Demonstration)

Stanley H. Roller, Chairman
U. S. Gypsum Company, Department 133, 101 S. Wacker Drive, Chicago, Illinois 60606

N1. Discovery of the world of sound. Stanley H. Roller (U.S. Gypsum Company, Department 133, 101 S. Wacker Drive, Chicago, IL 60606)

This is an audiovisual demonstration of a wide range of acoustical phenomena originally prepared to instruct architects on the physical phenomena important to their work. Among the topics presented are transmission loss, binaural hearing, the precedence effect, etc. The presentation is approximately one hour in length and will be shown at 9:00, 1:00, 3:00, and 5:00.

WEDNESDAY MORNING, 14 MAY 1986 RITZ ROOM, 8:45 TO 10:30 A.M.

Session O. Noise III: Radiation, Diffraction, and Transmission

Larry H. Royster, Chairman
Department of Mechanical & Aerospace Engineering, North Carolina State University, Raleigh, North Carolina 26769-7910

Contributed Papers

8:45

O1. Acoustic propagation under lapse conditions using fast Fourier transforms. S. R. Lloyd and W. K. Van Moorhem (Mechanical and Industrial Engineering Department, University of Utah, Salt Lake City, UT 84112)

In this paper the propagation of acoustic waves from a point source above a finite impedance ground under a temperature lapse condition is analyzed. The temperature profile chosen is a model of a fully developed lapse condition that fits the available data. This profile has a large temperature gradient near the ground and asymptotically approaches a finite value high above the ground. An analytical solution of the wave equation with variable sound speed is obtained using Hankel transform techniques. The present study implements a fast Fourier transform technique to invert the analytical solution to real space. Sound pressure level results to horizontal distances of 1000 m have been computed. The results show the expected interference pattern within the insonified region. At distances of a few 100 m from the source a rapid sound pressure decline, indicative of the shadow boundary, is observed. The sound pressure level behavior predicted is in agreement with the limited data available. [Work supported by NASA.]

9:00

O2. Acoustic field of a finite array of out-of-phase simple sources. Adnan Akay (Mechanical Engineering Department, Wayne State University, Detroit, MI 48202)

Radiation field of a finite array of out-of-phase simple sources is obtained. The results are generalized for monopole, dipole, and axial quadrupole source arrays. Closed-form solutions are obtained for the sound power and radiation efficiency of the array. It is shown that the maximum acoustic intensity of an array of n sources occurs at the coincidence frequency and is proportional to n^2. Results for the radiation patterns and efficiencies are given for different array characteristics. [Work supported by NSF.]

9:15

O3. Acoustic radiation from bending waves of a plate. Adnan Akay (Mechanical Engineering Department, Wayne State University, Detroit, MI 48202) and K. Uno Ingard (Departments of Physics and Aeronautics and Astronautics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Acoustic radiation from bending waves of a plate is reexamined by considering radiation loading of a plate driven by a traveling harmonic force. The results show that the plate displacement vanishes at the coincidence frequency and the coincidence frequency depends on the phase velocity of the forcing function with respect to the speed of sound in the fluid. Viscothermal losses in the boundary layer are also considered in the derivations to examine the effects of these losses on the radiated sound pressure and conversion of mechanical energy into heat below the coincidence frequency. [Work supported in part by NSF.]

9:30

O4. Curved surface diffraction theory derived and extended using the method of matched asymptotic expansions. Allan D. Pierce, Geoffrey L. Main, and James A. Kearns (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Consideration is given to the top of a wide barrier with variable radius of curvature R. The surface has finite acoustic surface impedance Z. Because kR is assumed large, the illuminated region can be approximated by geometric acoustics, such that plane wave reflection rules apply locally. The intricate interference pattern between incident and reflected ray fields assumes a tractable analytical form near the barrier top, which is subsequently used in a MAE solution of the overall diffraction problem. Unambiguous length scales result for radial and tangential distances along the barrier top. The inner solution is developed by expressing the wave equation in terms of such scales, subsequently identifying the expansion parameter as $(kR)^{-1/3}$. A parabolic equation emerges, with a boundary condition involving a scaled impedance $(Z/\rho c)(kR)^{-1/3}$; the outer boundary condition results from matching to the geometric acoustics solution. Outer expansion of the solution of the parabolic equation into the shadow zone yields an inner boundary condition on the ray theory solution for the diffracted wave. Results are similar to those previously derived for electromagnetic diffraction problems by V. A. Fock, but the MAE interpretation facilitates an extension to problems of multiple barriers. [Work supported by NASA–Langley.]

9:45

O5. Barrier insertion loss with a directional sound source. J. B. Moreland (Westinghouse Electric Corporation, Research and Development Center, 1310 Beulah Road, Pittsburgh, PA 15235)

The purpose of the work described in this presentation is to investigate the effect of source directivity on the sound attenuation of an acoustical barrier. The approach consisted of forming the coherent summation of the sound pressures produced by the source and its image to calculate the sound pressure arriving at the barrier edge, and from that summation, calculate the mean-squared sound pressure arriving at the receiver. This approach accounts for interference phenomena observed when a barrier rests on a reflecting plane. The source directivity can have a considerable effect on the sound field. At a given distance from the source, differences as much as 10 dB were calculated and measured by orientating the major lobe of the source radiation pattern in directions which were between 0 and 90 deg with respect to the reflecting plane. Also, the interference due to ground reflections results in negative insertion loss at distances determined by the source, barrier, and receiver geometry, and the frequency. Finally, the orientation at which maximum insertion loss occurs depends on the distances between the source, barrier, receiver, and ground plane. These results are significant because they show that serious errors (as much as 20 dB) in predicting the insertion loss of a barrier in an otherwise reflection-free environment can result from neglecting the ground reflections.

10:00

O6. Sound transmission through foam-lined double panel constructions. J. Stuart Bolton and Edward R. Green (Ray W. Herrick Laboratories, Purdue University, West Lafayette, IN 47907)

This paper describes the sound transmission performance of foam-lined double panel constructions, in particular, the acoustical effect of two foam mounting arrangements. The foam considered is relatively stiff and partially reticulated, the type most often used in noise control. A recent theory is used to model the foam [J. S. Bolton and E. Gold, J. Acoust. Soc. Am. Suppl. 1 **77**, S59 (1985)]; the theory allows for both a frame wave and an airborne wave. The applicability of the theory is demonstrated by comparing theoretical and measured transmission coefficients and impulse responses for freely suspended foam layers. The extension of the theory to allow for bonded or unbonded facing panels is then described. It is shown that when the foam lining is bonded directly to the facing panels sound transmission through the foam occurs largely via the high impedance frame wave. As a consequence, the transmission loss of foam-lined double panels is improved, particularly at low frequencies, when the lining and panels are separated by a small air gap; in this case a larger fraction of the energy is carried by the more heavily damped airborne wave.

10:15

O7. Glazing sound transmission loss studies. Gregory C. Tocci, Timothy J. Foulkes (Cavanaugh Tocci Associates, Inc., Natick, MA 01760), and Randolph E. Wright (Monsanto Polymer Products Co., 800 North Lindbergh Boulevard, St. Louis, MO 63167)

Monsanto has recently commissioned sound transmission loss tests at Riverbank Laboratories on various glazing configurations, including single thickness, air spaced, and laminated glass. Comparisons of various parameters such as STC, glass thickness, total glazing thickness, and noise reduction performance for various noise sources are made. These data have also been used to assemble glazing cost versus STC relationships that are useful for the building design prefessions. The presentation will also include a comparison of laboratory test data versus field performance for some of the glazing systems. [Work supported in part by Monsanto.]

WEDNESDAY MORNING, 14 MAY 1986 — SAVOY ROOM, 8:15 TO 10:30 A.M.

Session P. Physical Acoustics IV: Nonlinear Acoustics and Acousto-Optics

Mack A. Breazeale, Chairman
Department of Physics, University of Tennessee, Knoxville, Tennessee 37996-1200

Contributed Papers

8:15

P1. Nonsymmetric effect in finite amplitude sound beams radiating from a baffled circular transducer. H. C. Miao (General Motors Research Laboratories, Warren, MI 48090-9055) and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30338)

Prior investigations of nonlinear effects in sound beams have treated cases where the transducer oscillates axisymmetrically. Here, an analysis of a situation where the harmonic spatial vibration of the transducer has a cos θ dependence on the azimuth angle, as would be the case for a piston that rocks about its diameter, shall be presented. The method of investigation parallels that employed earlier [H. C. Miao and J. H. Ginsberg, J. Acoust. Soc. Am. Suppl. 1 **78**, S39 (1985)], which used the King integral to generate nonlinear source terms. A dual asymptotic description based on assumptions appropriate to the regions very close to, and far from, the beam axis is obtained, and then reconciled to obtain a uniformly accurate description. An intermediate form of the solution featuring coordinate straining transformations is converted to a Fourier time series. The linearized signal shows nodal lines in the azimuthal direction that match those of the transducer vibration, and it shall be shown that the higher harmonics exhibit similar behavior. [Work supported by ONR, Code 425-UA.]

8:30

P2. Relationship between near and farfield effects in second harmonic generation in the piston beam. M. A. Foda and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

An earlier analysis [J. H. Ginsberg, J. Acoust. Soc. Am. 76, 1201–1214 (1984)] of the infinite baffle problem for an axisymmetric harmonic excitation derived a nonlinear King integral describing signal distortion in sound beams. That analysis, which considered only the contribution to the cumulative distortion process associated with secular source terms, agreed with experiments at high frequencies at ranges beyond approximately one quarter the Rayleigh length. However, there was significant discrepancy with measurements very close the transducer face. Such differences will be shown to result from neglecting nonsecular second-order terms that do not carry into the farfield. In the present work, the nonsecular contribution to the second harmonic is evaluated as a numerical inversion of a multiple integral transform for the second-order potential. The composite effects of secular distortion, which is calculated using the renormalized King integral, and the nonsecular contribution are calculated. The results obtained from the analytical model compare favorably with previoius nearfield measurements. [Work supported by ONR, Code 425-UA.]

8:45

P3. The nonpropagating hydrodynamic soliton in annular geometries. Erich Carr Everbach and Robert E. Apfel (Department of Mechanical Engineering, Yale University, 2159 Yale Station, New Haven, CT 06520)

The nonpropagating hydrodynamic soliton discovered by Wu *et al.* [Phys. Rev. Lett. 52, 16 (1984)] exists as a localized finite amplitude disturbance in a parametrically excited rectangular channel. The behavior of the nonpropagating soliton in annular resonators of various radii of curvature is investigated. The distortion of the soliton profile, which increases with decreasing radius of curvature, suggests that as energy is moved from lower to higher wavenumbers, the dispersion must increase to balance the increasing nonlinearity. At a critical radius of curvature, the soliton changes form. As the inner annular radius decreases further, the soliton merges with a nonlinear mode of the circular tank. This mode resembles the (1,2) normal mode in that it consists of a localized maximum–minimum pair, but occurs at a frequency slightly below that of the (0,2) normal mode. The behavior and properties of this nonlinear "sub-(0,2)" mode are compared to that of the one-dimensional nonpropagating soliton. [Work supported by ONR.]

9:00

P4. Numerical solution of Burgers' diffusion equation for arbitrary waveforms. R. H. Mellen (PSI Marine Sciences, New London, CT 06320)

By transformation of the functional argument, Burgers' nonlinear equation for plane waves in dissipative fluids reduces to the diffusion equation. An "exact" solution for an initially sinusoidal wave has been obtained [D. T. Blackstock, J. Acoust. Soc. Am. 36, 534–542 (1964)]. The advantage is that the waveform can be calculated for any range without iteration. Solutions are also possible for initial waveforms of arbitrary shape by using computer methods to determine the Fourier coefficients. Effects of phase reversal of sawtooth waves at a pressure-release surface are estimated. No simple solutions have been found for nonplane waves; however, effects of phase quadrature in refractive media can also be approximated by neglecting divergence and the local details of the caustic region. Waveform distortion, "*N*-wave" formation, migration of the spectral energy, and chaotic properties of random noise are illustrated.

9:15

P5. Observation of focal shift of a convergent beam: Another Rayleigh angle phenomenon. Peter B. Nagy,[a)] Kangra Cho, Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210), and Dale E. Chimenti (Materials Laboratory, Air Force Wright Aeronautical Laboratories, Wright–Patterson AFB, OH 45433)

The nonspecular reflection of acoustic beams at critical angles, especially at the Rayleigh angle, has been widely studied. An interesting detail, the focal shift of convergent ultrasonic beams, was recently predicted by Bertoni *et al.* [Trait. Sign. 2, 201–205 (1985)]. Their analysis shows that the axial displacement could be much higher than that of the better known lateral Schoch displacement. However, because the acoustic fields of well-collimated beams change much slower along the axis of the beam than in the laterial direction, this effect is less pronounced. It is shown that by carefully choosing the acoustical and geometrical parameters of a convergent beam, this otherwise rather weak effect is readily detectable. The cylindrically focused beam of a 10-MHz transducer of ϕ10-mm diameter and 100-mm focal length was reflected from a water–aluminum interface. It is shown by Schlieren photography, that much below, exactly at, and much above the Rayleigh angle there is no focal shift, while slightly below and above the Rayleigh angle the focal shift was found to be a few centimeters in negative and positive directions, respectively, which is in good qualitative agreement with theoretical predictions. [a)] Permanent address: Applied Biophysics Laboratory, Technical University, Budapest.

9:30

P6. Acoustically induced depolarization of light diffracted by surface acoustic waves. William R. Graver, Tran D. K. Ngoc, and Walter G. Mayer (Physics Department, Georgetown University, Washington, DC 20057)

Depolarization of light upon interacting with an interface is reviewed for two cases: with and without the presence of surface acoustic waves (SAW's). The characteristics of depolarization in the case of light diffraction by SAW's are examined oin the basis of a phenomenological model developed by Stegeman [G. I. Stegeman, J. Appl. Phys. 49, 5624–5637 (1978)]. This model was constructed by considering the two scattering mechanisms associated with the surface corrugation and elastoptic effects. Specifically, the *s*- and *p*-polarized electric field components are investigated as functions of the incident angle, acoustic intensity and acoustic frequency. [Work was supported by the Office of Naval Research.]

9:45

P7. The effect of laser beam oblique incidence and optical refraction on the characteristics of an optoacoustic source in a liquid. Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The generation of underwater sound by a laser beam is very attractive because the laser path can be accurately controlled by means of optical components (rotating mirrors, grating, lenses, etc.) so that a virtually unlimited number of source configurations are, in fact, achievable. In most practical designs, the laser beam illuminates at oblique incidence a body of water so that optical refraction actually occurs at the air–water interface. This effect is now investigated by extending the analysis presented previously [J. Acoust. Soc. Am. 78, 2074–2082 (1985)]. It is shown that the impulse response of a laser-induced thermoacoustic source is strongly dependent upon the angle of incidence of the laser beam, although optical refraction tends to minimize this effect. Directivity patterns and propagation curves are also found by taking the Fourier transform of the modified impulse response of the optoacoustic system; it is found that they can be significantly affected by an oblique incidence of the laser beam.

10:00

P8. Computerized Schlieren visualization. Peter B. Nagy,[a)] Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210), and Pal Greguss (Applied Biophysics Laboratory, Technical University Budapest, Budapest, Hungary)

In everyday scientific research, the sought information about the acoustical phenomenon under study is often conveyed by the beam profile, or more generally, by the spatial distribution of the acoustic field. In many cases, such as Rayleigh phenomena, diffraction, scattering, leaky surface wave, etc., the acoustic field of interest can be observed conveniently in an optically transparent media by Schlieren visualization. This method readily gives a qualitative picture of the overall phenomenon, but often we need more precise quantitative results. An easy-to-use laser based system which works in cw as well as pulsed operation up to a considerable speed of 15 ns. The real-time video display offers a convenient way of direct evaluation, but the whole picture or part of it can be digitalized and fed into a PDP 11/34 computer. Examples are presented showing how this technique can substitute troublesome hydrophone mapping by mechanical scanning, and how the accuracy and reliability of the result can be increased by such digital signal processing methods as time and/or spatial averaging, image subtracting, smoothening, contouring, etc. In addition, the cross section of axisymmetrical beams can be reconstructed by a simple computerized tomography technique from the measured projection. [a] Permanent address: Applied Biophysics Laboratory, Technical University, Budapest.

10:15

P9. Acoustic resonance of a cylindrical cell in an optoacoustic spectrometer. Wayne M. Wright, Edward A. Gardner (Department of Physics, Kalamazoo College, Kalamazoo, MI 49007), and Paul B. Hays (Space Physics Research Lab, The University of Michigan, Ann Arbor, MI 48109-2143)

The optoacoustic spectrometer is an instrument with which one can study the absorption of light by a sample of gas. Thermal energy provided by a periodically interrupted laser beam is converted to an acoustic signal of the same frequency, and the amplitude of this signal often is increased through operation at an acoustic resonance frequency of the sample cell. The construction of one such spectrometer and its use to study the variation of acoustical properties of a cylindrical cell with ambient pressure are described. Measurements have been made with the monochromatic light produced by a tunable dye laser, with atmospheric oxygen as the optically absorbing gas, and over the pressure range 0.05–1.0 atm. The acoustic "Q" for the 4.1-kHz first azimuthal mode of the 5-cm-diam cylindrical sample cell is found to decrease by less than 10% (from approximately 100) as the pressure is reduced from 1 to 0.5 atm, and then to fall off much more rapidly. The position of the commercial 1-in. condenser microphone on the cylindrical cell wall is shown to have a significant effect on the cell resonance frequency. [Work supported, in part, by NSF.]

WEDNESDAY MORNING, 14 MAY 1986 — WEST BALLROOM, 8:00 TO 10:00 A.M.

Session Q. Physiological Acoustics III and Psychological Acoustics IV: Processing of Stimulus Level

Sid P. Bacon, Chairman
Boys Town National Institute, 555 North 30th Street, Omaha, Nebraska 68131

Contributed Papers

8:00

Q1. The external ear sound pressure level transformation in infants. Richard S. Bernstein and Barbara Kruger (Department of Otolaryngology, Albert Einstein College of Medicine, Bronx, NY 10461)

Some preliminary findings obtained in a study of the acoustic characteristics of the external ears of infants are reported. A technique was developed for positioning the inlet of a probe-tube microphone in the lateral-half of the ear canals of sleeping infants. A diffuse sound field (spectral density approximately 42 dB SPL) was introduced and the microphone output recorded. Free-field-to-ear canal sound pressure level transformations were determined for infants ranging in age from a few days old to 36 months. Representative transformations are presented, and these vary systematically with the age of the child. The resonance frequency of the external ear is high in newborns and declines exponentially with age. The asymptotic value (approximately 2700 Hz) appears to be reached during the second year of life. [Work supported by NIH-NINCDS.]

8:15

Q2. Design of prostheses for the middle ear. A. M. Nassef, R. D. Finch, F. Mistree (Department of Mechanical Engineering, University of Houston, Houston, TX 77004), and L. Gray (Department of Otolaryngology—Head and Neck Surgery, University of Texas Medical School, Houston, TX 77030)

The middle ear has been modeled with an assembly of mechanical components, and the values of the parameters in that model were determined using an optimization technique [J. Acoust. Soc. Am. Suppl. 1 77, S94 (1984)]. Certain diseases of the middle ear are treated by replacement of the three ossicles which transmit sound from the outer ear to the inner ear. It is normal to replace the three ossicles by a prosthesis consisting of a simple rod. Such devices do not restore hearing fully, and the same device is not suitable in all cases. The overall aim of this work is to develop the capability to select the best configuration to restore acoustical response, to choose the best material to be used in manufacturing such a prosthesis, and to determine the dimensions of the device. To reach this aim, the following are the steps to be taken: (1) Reproduce the acoustical transfer function of a normal middle ear as accurately as possible; and (2) meet surgical constraints such as ease of implantation and resistance to disease. In this paper, we discuss some designs, set up the mathematical formulation to determine the principal dimensions of these different designs, and present some results which include a comparison of the characteristics of new designs and those currently used.

8:30

Q3. The role of discharge history effects in rapid adaptation of auditory-nerve fibers. Bernd Lutkenhoner and Robert L. Smith (Institute for Sensory Research, Syracuse University, Syracuse, NY 13210)

PST histograms reflecting the response of auditory-nerve fibers to high-intensity bursts show a high initial peak followed by an exponential decay with a time constant of a few milliseconds ("rapid adaptation"). The role discharge history effects play in this phenomenon is being investigated. It is assumed that the probability for a spike in the infinitesimal interval $(t,t+dt)$ is given by the expression $s(t)r_1(x_1)...r_m(x_m)\,dt$, where s is the "excitation function," r_i is the ith order recovery function, and $x_i=x_i(t)$ is the time since the ith prior discharge $(i=1,2,...,m)$. Higher-order recovery functions have been suggested by Gaumond [Doctoral thesis, Washington University, St. Louis, MO (1980)] who also demonstrated the existence of the second-order function r_2. Our experimental investigations confirm his results and furthermore reveal third- and fourth-order effects. Monte Carlo simulations show that the first-order recovery term can explain only the initial peak of the rapid adaptation phenomenon. In contrast, higher-order effects can contribute substantially to the exponential decay. [Work supported by the Deutsche Forschungsgemeinschaft, NSF and NIH.]

8:45

Q4. Extraction of the neural count function from the Weber fraction. William S. Hellman (Department of Physics, Boston University, Boston, MA 02215) and Rhona P. Hellman (Department of Psychology, Northeastern University, Boston, MA 02115)

It is shown that the dependence of the Weber fraction, $\Delta I/I$, on I determines the form of the neural firing rate function $N(I)$ provided that the relation between ΔN and N is given. Here, ΔN is the change in average firing rate resulting from an intensity jnd. The firing rate function consistent with Weber's law is predicted when $\Delta N \sim N^{1/2}$. Assuming that loudness is proportional to the neural firing rate, loudness equations are constructed for two stimuli where Weber's law is obeyed and empirical loudness functions are available. These stimuli are (1) a 100-Hz tone partially masked by an adjacent high-pass noise and (2) broadband noise. Close agreement between the derived and measured loudness functions is obtained. The procedure also predicts a power law for loudness, with a generalization, when the near miss to Weber's law is represented by a power function. [Partially supported by the Rehabilitation Research and Development Service of the VA.]

9:00

Q5. Intensity jnd's for equally loud tones in quiet and noise backgrounds. C. M. Rankovic, M. F. Cheesman (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455), D. A. Fantini, C. Uchiyama, and N. F. Viemeister (Department of Psychology, University of Minnesota, Minneapolis, MN 55455)

Recent studies of the relation between loudness and intensity jnd's suggest that the size of the jnd corresponds to loudness magnitude rather than to the slope of the loudness function. Specifically, Zwislocki and Jordan [J. Acoust. Soc. Am. Suppl. 1 77, S64 (1985)] conclude that the intensity jnd's for a pure tone are equal in recruiting and normal ears when the loudnesses of the tones are equal. Similarly, Hellman *et al.* [J. Acoust. Soc. Am. Suppl. 1 77, S64 (1985)] conclude that when a pure tone in narrow-band noise is judged to be equally loud to a tone in wideband noise, the jnd's for those tones are the same. We show exception to these studies. Loudness matches and intensity jnd's for a 1000-Hz pure tone in quiet and in a 40-dB spectrum level broadband noise were obtained for four normal-hearing subjects. The data indicate that equally loud tones yield equal jnd's only at intensities near threshold and at high intensities where equal loudness corresponds to equal SPL. At other intensities, the jnd is larger in noise than in quiet. [Supported by NIH NS12125, NS07889, and by a training grant, NICHHD T32 HD-07151, awarded to CMR by the Center for Research in Human Learning.]

9:15

Q6. Level discrimination as a function of level and frequency between 0.25 and 16 kHz. Mary Florentine (Communication Research Laboratory, 133 FR, Northeastern University, Boston, MA 02115), Soren Buus (Psychology Department, 282 NI, Northeastern University, Boston, MA 02115), and Christine R. Mason (Communication Research Laboratory, 133 FR, Northeastern University, Boston, MA 02115)

Difference limens, DLs, for the level of tones were measured at 0.25, 0.5, 1, 2, 4, 8, 10, 12, 14, and 16 kHz as a function of level. Approximately ten levels ranging from near threshold to about 100 dB SPL were used. The stimuli were 500-ms tones with an interstimulus interval of 250 ms. An adaptive 2*I*, 2AFC procedure with feedback was used. The results show rather large consistent individual differences among normal listeners, but some patterns can be observed in the average DLs. First, DLs at all frequencies are generally better at high levels than at low ones, but the DLs above 8 kHz often are larger at moderate levels than at high and low levels. Second, the DLs at any fixed SPL are largely independent of frequency up to 4 kHz, but increase with frequency above 4 kHz. The behavior of the DLs at high frequencies presents a challenge to current models of level discrimination. [Supported by NIH-NINCDS RO1NS18280.]

9:30

Q7. Effect of sloping high-frequency cochlear losses on loudness functions at lower frequencies. Rhona Hellman and Carol Meiselman (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

The relation between loudness at low and middle frequencies and auditory thresholds at high frequencies was determined in a hearing-impaired population with sloping high-frequency cochlear losses. Nine people, all with steep symmetrical bilateral losses, participated in the study. Loudness was measured at two frequencies in the normal-hearing region by magnitude estimation and production. Despite a marked increase in thresholds at frequencies just above the upper test frequency, both the dynamic range measured for loudness and the tone's rate of loudness growth were essentially the same at the two test frequencies. Moreover, the loudness functions were the same as the monaural loudness function measured at 1000 Hz in normal hearing. Consistent with the findings of other studies, the results indicate that the magnitude of loudness does not depend on the high-frequency excitation pattern evoked by the tone in the ear. [Supported by the Rehabilitation Research and Development Service of the VA.]

9:45

Q8. Four factors in loudness change. Laurie Smith and Ernest M. Weiler (Department of Communication, Mail Location #379, University of Cincinnati, Cincinnati, OH 45221)

The use of direct magnitude estimation (ME) of loudness has recently been very useful in the study of auditory adaptation effects. The current study proposed to compare ME loudness adaptation to the widely studied adaptation effects measured by the classic simultaneous dichotic loudness balance technique. Three conditions were administered to each of 18 SS at 60 dB SPL, 1000 Hz. Condition I consisted of 7 min of the stimulus with ME judgments taken at the beginning and end. Condition II consisted of the SDLB procedure with binaural pretest balance values determined in dB, 7 min of monaural stimulation in the second stage, and dB balance determined binaurally in the final stage. ME loudness values were also taken at each stage. Condition III consisted of a parallel to condition II, but without the dB adjustments or balances that are normally found in SDLB procedures. Only ME judgments were taken during the three stages of condition III. Results of ten dependent measures were subject to factor analysis, with four factors identified. These independent factors appear to account for the reported disagreements between ME loudness adaptation and SDLB related effects.

Session R. Shock and Vibration II: Acoustic Intensity—Structural Acoustics

Albert Tucker, Chairman
Code 474, Office of Naval Research, 800 North Quincy Road, Arlington, Virginia 22217

Chairman's Introduction—8:30

Contributed Papers

8:35

R1. Acoustic energy transfer and intensity vortices. J. Adin Mann, III and Jiri Tichy (Graduate Program in Acoustics, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

Concepts of energy transfer in an acoustic field will be discussed using the vectorial, transient, and steady-state properties of acoustic intensity. Emphasized is the region of an active intensity vortex. Pascal's decomposition of the active intensity into a part with zero and nonzero curl will also be included, but with different conclusions. Calculated results of the build up of a vortex and steady-state fields and data from an automated measuring system will be used in the discussion.

8:50

R2. Structural intensity flow in and acoustic radiation from thin plates with damped structural discontinuities. Karl B. Washburn (Sachs/Freeman Associates, Inc., 1401 McCormick Drive, Landover, MD 20784) and Earl G. Williams (Naval Research Laboratory Code 5133, Washington, DC 20375-5000)

SIMMAP (structural intensity mapping from measurement of acoustic pressure), an outgrowth of nearfield acoustical holography, provides two-dimensional mappings on the surface of a vibrating plate of the structural intensity, mechanical injected power, and acoustic power radiated into the fluid [E. G. Williams *et al.*, J. Acoust. Soc. Am. **78**, 2061–2068 (1985)]. The relationship between mechanical injected power (into the plate), radiated acoustic power (from the plate), and the effects of structural discontinuities on both are demonstrated over the surface region of the plates. The data from the four experimental configurations confirm that, while acoustic intensity mapping effectively pinpoints radiation sources, SIMMAP is more dependable in locating mechanical sources (e.g., a shaker) and sinks (e.g., damped structural discontinuities). The technique clearly reveals net mechanical power extraction by damped discontinuities and preferential acoustic radiation from regions where they are attached, indicating a powerful tool for analysis of vibration damping. [Work supported by DTNSRDC, Annapolis, MD.]

9:05

R3. A scanning microphone implementation of nearfield acoustical holography. William Y. Strong, Jr. and Gordon Ebbitt (CBS Technology Center, 227 High Ridge Road, Stamford, CT 06905)

A system that implements the nearfield acoustical holography (NAH) technique in air has been designed and placed into operation at the CBS Technology Center. The system utilizes a scanning microphone that may be positioned within a $3.7\times3.7\times1.2$ m volume by a computer controlled robot. Sound pressure data from the scanning microphone and a reference are stored in a mini computer and subsequently processed using nearfield holographic techniques. The resulting reconstructions of the pressure, velocity, and vector intensity fields may be displayed using either a 3-D video graphics system or a digital plotter. A detailed discussion of this NAH system will be presented and reconstructions of the sound field generated by a simple source will be shown.

9:20

R4. Vibrational characteristics of an infinite flat plate with varying properties. Mauro Pierucci and Tuan Le (Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

A theory to study the vibrational characteristics of a structure with properties varying along its length is developed. The analysis consists of assuming that the plate thickness and plate stiffness are functions of the axial distance x. The resulting thin plate equation becomes a fourth-order differential equation with variable coefficients. The equation is solved by Fourier transform techniques and a solution in the transformed plane is obtained. The analysis is then applied to a constant thickness plate with sinusoidally varying stiffness. For simplicity, the effect of fluid loading is neglected and the forcing function is assumed to be given by a distributed load. Results in the form of plate displacement disribution are presented.

9:35

R5. Wavenumber/frequency response of free beams to multiple-point excitation. Frederick M. Hutto, Courtney B. Burroughs, W. Jack Hughes, and Karl Grosh (Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

Using Timoshenko beam theory, an analytic model for the free response of a thick free–free beam and the forced response of the beam to a continuous drive and an array of point drives is presented. To validate the model, comparisons of predictions are made to measurements of the free response of a beam, the magnitude and phase of the point and transfer admittances of a beam driven by point drives of one, two, and three locations, and the wavenumber/frequency response of a beam driven at point drives at one, two, and three locations. The excellent agreement shown validates the analytic model, which is then used to show the wavenumber response of a beam as a function of the wavenumber of a continuous drive and the steered wavenumber of an array of point drives.

9:50

R6. Application of a variational principle to acoustic radiation from a vibrating finite cylinder. X.-F. Wu, A. D. Pierce, and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A variational principle derived from the Kirchhoff–Helmholtz integral corollary is applied to a rigid cylinder; previous applications have been to acoustic radiation and diffraction from an infinitely thin disk [J. H. Ginsberg *et al.*, J. Acoust. Soc. Am. Suppl. 1 **77**, S60 (1985)]. For a finite length circular cylinder, additional complications are encountered because of the need to consider the cylindrical sides as well as ends. The basis functions for a Rayleigh–Ritz solution have different analytical forms on the sides and ends, although they must be continuous at the edges. A larger number of functions are required to get a good representation of the surface pressure. Calculation of the resulting matrix elements for the determination of the expansion coefficients is nontrivial because of the logrithmic singularities presented by the Green's function integrals.

The surface pressure is expected to be finite, but the tangential derivative of the surface pressure is *a priori* known to have a $s^{-2/3}$ singularity at the edges separating the side wall and ends. It is not necessary that trial functions exhibit the latter behavior, but if they are so chosen, a good convengence is expected for a smaller number of basis functions. A numerical method based on splitting each integrand into a simple singularity term plus a nonsimple but bounded term is described. Numerical results are compared with those obtained by an alternative technique developed by P. H. Rogers [NRL Report 7240 (19 June 1972)].

10:05

R7. Comparison of different methods for the dynamic analysis of beam surfaces. Massoud S. Tavakoli (Department of Mechanical Engineering, The Ohio State University, 206 W. 18th Avenue, Columbus, OH 43210)

Beam structures such as multispan beams and frames are used to compare the dynamic analysis methods. The first method is the receptance technique in which the precalculated receptance coefficients are used to model the junction of the beam segments. The second method is concerned with the component mode synthesis where simple polynomials are utilized as mode shapes. Some or all of the joint conditions are then enforced and hence the natural frequencies and mode shapes are approximated. The third method takes advantage of the energy approach. The Lagrangian for the entire system is obtained by adding the Lagrangians of individual components. Minimization of the overall Lagrangian results in the natural frequencies and mode shapes. Finally, the advantages and disadvantages of these methods are discussed.

WEDNESDAY MORNING, 14 MAY 1986 — EAST BALLROOM, 8:00 TO 10:30 A.M.

Session S: Speech Communication IV: Production: Various Topics (Poster Session)

Robert J. Porter, Jr., Chairman
Department of Psychology, University of New Orleans, New Orleans, Louisiana 70148

All posters will be displayed from 8:00–10:30 A.M. To allow all contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00–9:15 A.M. and contributors of even-numbered papers will be at their posters from 9:15–10:30 A.M.

Contributed Papers

S1. Stress pattern differentiation of form class in English. Joan A. Sereno (Box 1978, Linguistics Department, Brown University, Providence, RI 02912)

The present experiment explores the regular stress pattern that differentiates nouns and verbs in English. In some English words, a shift in stress from the first to the second syllable produces a change in form class from noun to verb (e.g., SURvey, noun; surVEY,verb). However, preliminary results suggest that this stress pattern is not solely restricted to this rather limited set of words. An analysis of the Brown Corpus [Francis and Kučera, *Frequency Analysis of English Usage* (Houghton Mifflin, Boston, 1982)] revealed that the vast majority (92%) of bisyllabic words used only as nouns in English have first syllable stress (e.g., "student," "country") whereas the great majority (85%) of bisyllabic verbs have second syllable stress (e.g., "begin," "receive"). In the present experiment, "ambiguous" bysyllabic words (i.e., words that do not change their stress placement with a change in form class) were examined. In these words (e.g., "answer," "design"), acoustic measurements (duration, amplitude, fundamental frequency, and second formant frequency levels) were taken. Some stress-related acoustic differences were found between the ambiguous bisyllabic words read as nouns and those same words read as verbs. The results will be discussed in terms of their significance for lexical organization.

S2. Effects of stress and final consonant voicing on vowel articulation and formant patterns. Walter V. Summers (Department of Psychology, Indiana University, Bloomington, IN 47405)

Stress level and voicing of a postvocalic consonant are two factors known to influence vowel duration. In the present study, vowel durations, jaw and lower lip movement data, and formant data were examined for a set of b–vowel–consonant utterances differing in stress and final consonant voicing. As expected, vowel durations were greater for stressed utterances and for utterances containing voiced final consonants (voiced utterances). Articulator lowering and raising gestures were more rapid and more extensive for stressed versus unstressed utterances. At the acoustic level, stressed uterances demonstrated more rapid initial formant transitions and more extreme steady-state frequencies than unstressed utterances. In contrast, increases in vowel duration due to voicing were not accompanied by increases in the velocity or magnitude of articulatory gestures. In addition, voiced utterances did not demonstrate steeper formant transitions or more extreme steady-state frequencies than voiceless utterances. The results demonstrate that stress-related and voicing-related changes in vowel duraton are accomplished by separate and distinct transformations with observable consequences at both the articulatory and acoustic level. [Work supported by NINCDS and the Stuttering Center, Baylor College of Medicine.]

S3. Interaction of phonemic quantity with speech rate and emphasis. Riitta Välimaa-Blum (Department of Linguistics, Ohio State University, 1841 Millikin Road, Columbus, OH 43210)

Finnish has a quantity contrast in both consonants and vowels. It also has primary stress on the word-initial syllable. But, unlike in Czech, there is no neutralization of vowel length in stressed positions, i.e., a stressed syllable may contain either a phonemically short or long vowel. Vowel length is one of the major stress cues in a language like English but obviously in Finnish this use would conflict to some extent with its use in contrasting phonemes. In comparing word structures CVCV, CVVCV, and CVCCV in Finnish it was found that the final vowel in CVCV words was significantly longer than in the other two where the first syllable is long. It would appear as if the effect of the stress were spread over the two syllables in the CVCV words. The same lengthening has been found in an earlier study in normal speech rates in nonemphatic positions. The present study found this effect in both normal and fast speech rates, in two different sentence positions and in three different emphasis conditions.

S4. Tone and intonation in Cantonese. Keith Johnson (Department of Linguistics, Ohio State University, Columbus, OH 43210)

The phenomenon known as an $F0$ downtrend has recently been attributed to local phonological rules (such as downstep and final lowering) in English. A preliminary study of Cantonese seemed to indicate that a backdrop $F0$ decline is an important aspect of the Cantonese intonational system. In that study it was found that the preservation of lexical tones in Cantonese sentences could best be accounted for when declination was included in the description. In the study to be reported in this paper the method used in the preliminary investigation is refined and two tests of a possible declination effect are employed. The refinements include: (1) recording sentences in utterance medial position, and (2) using pragmatically neutral sentences (i.e., the target sentences neither introduce nor conclude a discourse topic). The first test for declination is the tone preservaton criterion used in the preliminary study. The second has to do with the presence or absence of a correlation between the ratio of $F0$ measurements of early and late occurrences of a tone and the number of syllables intervening between the two occurrences. Such a correlation between $F0$ ratio and number of intervening syllables would indicate a declination effect (if the slope is negative). If no evidence for declination is found then the approach which attributes downtrend to local phonological rules is supported. If evidence for declination is found then declination must be included among the possible methods of accounting for downtrend.

S5. Patterns of coarticulation in English. Marie K. Huffman (Department of Linguistics, University of California, Los Angeles, CA 90405)

Recent work on vowel-to-vowel coarticulation [e.g., Krakow and Manuel (1984), Magen (1984)] has identified language specific coarticulatory patterns. These include differences in relative strength of carryover versus anticipatory coarticulation (sometimes called "directionality") and differences in the freedom of vowels to vary. Such results suggest that coarticulatory patterns of individual language are more complicated than was thought previously. The present study examines vowel-to-vowel coarticulation in English with respect to several variables, primarily stress and consonant manner of articulation. Tokens recorded were of the form VCV and bVCəCVb, where the vowel was one of /i,a,u/ and the consonant was one of /1,d,r/. These nonsense words were said in a frame, with stress on the first or the last syllable. Formant frequencies were measured, using LPC analysis, at several points in these tokens. Our data suggest that stress of the affected vowel influences amount of carryover coarticulation more than stress of the affecting vowel does. Other results include indications that coarticulatory directionality can vary within a single language; here both identity of the consonant and stress play a role. [Work supported by NSF.]

S6. Coarticulation across secondary articulations. Patricia A. Keating (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

Consonants with secondary articulations, which involve simultaneous consonant and vowel-like articulations, are found as contextual allophones in many languages and as independent phonemes in some languages. These consonants show special behavior with respect to vowel-to-vowel coarticulation, as determined from acoustic measurements. Phonemic secondary articulations in stops, fricatives, and liquids of Russian, Bulgarian, and Arabic generally block any vowel-to-vowel influence across the consonant. However, contextual secondary articulations in Russian, Bulgarian, Polish, and English generally propagate extreme effects of one vowel onto the other. Both of these differ from the usual coarticulation across consonants without secondary articulations. These results can be accounted for if we distinguish the representations of phonemic secondary articulations (as consonants with their own vowel features), from contextual secondary articulations (as consonants acquiring features from adjacent vowels), from lack of secondary articulation (as consonants with no vowel features). Given such a distribution of vowel features, phonetic interpolation can proceed straightforwardly between adjacent feature values.

S7. Long consonants versus geminate consonants in Arabic. Ann M. Miller (Linguistics Department, 204 Cunz Hall, 1841 Millikin Road, The Ohio State University, Columbus, OH 43210-1229)

If a phonetic distinction can be made between long consonants and geminate consonants (a cluster of two like consonants), it should be cued by durational differences in the absence of a distinctive release. Arabic provides a means for comparing these types of consonants since it has phonemically distinct long consonants word initially and medially, and clusters of like consonants across word boundaries. We performed an experiment with speakers of Levantine Arabic to compare these conditions. We also tested clusters which result from assimilation of the /1/ in the definite article /ʔil/ to following dental consonants to see whether these patterned like true clusters or like long consonants, indicating cliticization. Surprisingly, no differences in duration occurred between these conditions. Moreover, one occurrence of a medial long consonant had an apparent release, as did several of the clusters across word boundaries and one of the clusters involving the definite article. These results suggest that medial lengthened consonants in Arabic are not long consonants as in Japanese and Finnish but are actually clusters.

S8. Declination in Toishan. Gina M. Lee (Department of Linguistics, The Ohio State University, 204 Cunz Hall, 1841 Millikin Road, Columbus, OH 43210)

Many have noted a tendency for fundamental frequency to gradually lower during the production of an utterance. This downtrend has sometimes been attributed to *declination*, a phonologically unmotivated lowering of $F0$ that is independent of tone. Recently, Johnson [Meeting Handbook, LSA, 21 (1985)] discovered data consistent with the view that declination exists in Cantonese. Like Cantonese, Toishan (another Yue dialect) exhibits little, if any tone sandhi. To see whether Toishan likewise exhibits declination, we had native speakers produce sentences containing high level tones, with varying amounts of material between the target high tones. The sentences are organized into passages so that, in addition to an examination of declination at the sentence level, an examination can be made at the discourse level as well (since it is possible that the degree of declination within a sentence depends on the position of the sentence within the discourse unit).

S9. Palatoglossus activity during oral/nasal vowels of Hindi. R. Prakash Dixit (Department of Speech: Communication Disorders, Louisiana State University, Baton Rouge, LA 70803), Fredericka Bell-Berti (Department of Speech Communication and Theatre, St. John's University, Jamaica, NY 11439 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510), and Katherine S. Harris (The Graduate Center, CUNY, New York, NY 10036 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Palatoglossus EMG potentials and voice were recorded while a native speaker of Hindi produced CVC nonsense utterances where C was /s/ and V was one of the vowels /i ɩ e ɛ ə ɑ ɔ o ɷ u, ĩ ɩ̃ ẽ ɛ̃ ə̃ ɑ̃ ɔ̃ õ ɷ̃ ũ/. Preliminary results show low levels of EMG activity for the front oral vowels, whereas for the central and back oral vowels the EMG activity is considerable. All nasal vowels exhibit a high level of EMG activity, except /ẽ, ɛ̃/ for which the EMG activity is minimal. EMG levels for the central and back nasal vowels are considerably higher than those for the front nasal vowels. EMB levels for the nasal vowels /ɔ̃, õ, ɷ̃, ũ/ are comparable to those for the oral vowels /ɔ, o, ɷ, u/, respectively. However, the EMG levels for the nasal vowels /ĩ, ɩ̃, ə̃, ɑ̃/ are considerably greater than for the comparable oral vowels /i, ɩ, ə, ɑ/. Furthermore, the patterns of EMG potentials for the long vowels differ from those for the short vowels irrespective of their being oral or nasal. [Work supported by NIH Grant NS-13617 and BRS Grant PR-05596 to Haskins Laboratories.]

S10. Articulatory reorganization in deaf talkers. Mary Joe Osberger (Department of Communicative Disorders, University of Wisconsin, Madison, WI 53706), Ronald Netsell (Boys Town National Institute, Omaha, NE 68131), and David Goldgar (Division of Biostatistics, University of Mississippi, Jackson, MI 39216)

The ability of five intelligible deaf talkers to reorganize articulatory gestures was examined with the use of a bite block. Each subject produced the phrase "say a pVp again" (where V = [i, a, u, æ, ɪ] or [ɛ]), with 14 repetitions of each vowel in random order. The speech samples were obtained with and without the subjects wearing their hearing aids. Measures of VOT, vowel duration, F_1, and F_2 were obtained from wideband spectrograms. The speech of the two subjects with an early onset of deafness was affected by the presence of the bite block whereas the speech of the remaining three subjects, whose deafness occurred in adulthood, was not. Performance was similar for all subjects irrespective of whether they were wearing their hearing aids. The role of auditory information in the formative years of speech motor control is discussed. [Research supported by the Easter Seal Research Foundation and a Biomedical Research Support Grant from the NIH.]

S11. Correlation of SPINE test scores with measures of tongue deviancy computed from formant frequencies in hearing-impaired speakers. Donald C. Wold (Department of Physics and Astronomy, University of Arkansas at Little Rock, 33rd and University Avenue, Little Rock, AR 72204), Christy R. Evans, Jesse E. Dancer, and James C. Montague (Department of Communicative Disorders, University of Arkansas at Little Rock, Little Rock, AR 72204)

When the first three formant frequencies are given in vowel production, plausible midsagittal vocal tract shapes may be generated easily [D. C. Wold, J. Acoust. Soc. Am. Suppl. 1 **78**, S55 (1985)]. As a measure of tongue deviancy, this technique was used to compute the root-mean-square deviation with vocal-tract widths computed from measured formant frequencies and the standard widths computed from average formant frequencies for adults. The vowels selected for analysis were /i/,/u/, and /a/. A perceptual speech intelligibility test for the hearing-impaired, called SPINE for speech intelligibility evaluation, was administered to 28 hearing-impaired speakers. The Pearson product-moment correlation was computed between the SPINE test scores of the subjects and the inverse of the tongue deviancy. The preliminary results were positive with r equal to 0.53 ($p<0.01$), 0.02, 0.27, and 0.54 ($p<0.01$) for /i/, /u/, /a/, and the vowels combined, respectively. Thus verbal speech intelligibility scores for the hearing-impaired group were inversely related to a simple measure of tongue deviancy.

S12. Lip–jaw coordination in hearing-impaired speakers. N. S. McGarr, R. A. Seider, and A. Löfqvist (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The question posed is whether interarticulator timing relationships are preserved in deaf speakers across speech transformations such as those induced by phonetic context and stress. Acoustic recordings were obtained simultaneously with kinematic records of lip and jaw motions using the Selspot infrared tracking device. The experimental corpus included real word utterances with a medial labial consonant (p,b,m,w,f,v) flanked by one high and one low vowel (e.g., "And Bea pops it"). The stimuli were produced by one hearing and three hearing-impaired talkers. Measures of jaw and lip displacement and velocity of movement were made in order to address questions of kinematics as a function of stress, phonetics, and vowel effects. Results for the hearing speaker showed the expected systematic differentiation of stressed and unstressed syllables. Preliminary results for the hearing-impaired speakers showed the relative interarticulator timing was more variable than the normal. However, the deaf speakers were like normal in that they differentiated vowels on the basis of jaw placement. These and other measures to investigate phonetic effects will be presented. [Work supported by NIH Grants NS-13617 and NS-13870.]

S13. Speech produced under high sustained acceleration: Some preliminary observations. Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701), Thomas J. Moore, and Timothy R. Anderson (Aerospace Medical Research Laboratory, Wright–Patterson AFB, OH 45433)

The purpose of this study was to obtain preliminary data concerning the acoustic-phonetic structure of speech produced under high sustained acceleration. Acoustical measurements were made of a set of Air Force vocabular words as spoken by two subjects at 1 g and 6 g's. There were differences in both the durational and spectral characteristics of speech, though not always consistently for the two speakers. At 6 g's, vowel formants tended to centralize, and fundamental frequency in stressed syllables increased for both speakers. For one speaker, word durations increased consistently under acceleration while for the other speaker, the durational differences were inconsistent. Duration differences were primarily a function of changes in vowel duration. [Sponsored in part by the Air Force Office of Scientific Research/AFSC, United States Air Force, under Contract F49620-85-C-0013.]

S14. Formant estimation of high fundamental frequency speech. Corine Bickley (Room 36-521, Department of Electrical Engineering and Computer Science and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Formant measurement procedures often rely on there being a low fundamental frequency. An early study [B. Lindblom, International Congress of Phonetic Sciences, 4th, Helsingfors, **1961**, 189–202 (1962)] found that the mean error in formant estimation ranged from about 40 Hz to a frequency of one-fourth the fundamental. This study compares signal processing techniques for the estimation of formant frequencies and bandwidths of synthesized and natural speech characterized by a high fundamental frequency. Utterances were synthesized [D. H. Klatt, J. Acoust. Soc. Am. 67, 971–995 (1980)] using young children's utterances as models. The spectral and durational characteristics were matched closely by manipulating the synthesizer parameters. Spectrograms, discrete Fourier transforms, linear prediction envelopes, and auditory pseudospectrograms were computed for both the synthesized and natural utterances. The accuracy of formant estimation was judged by comparing the values determined by each of these methods to the known frequencies and bandwidths of the synthesized speech. Implications for formant estimation of natural speech will be discussed. [Work supported in part by a Whitaker Health Sciences Fellowship.]

S15. The effects of contrastive stress on stutterers' speech. G. V. Klouda and W. E. Cooper (Department of Psychology, University of Iowa, Iowa City, IA 52242)

Acoustic analyses were performed on the utterances of nine stutterers to examine the influence of sentence focus on frequency of stuttering and fundamental voice frequency ($F0$) patterns. The speakers orally read sentences containing contrastive stress prompted by preceding questions. The application of focus on a word led to a statistically significant increase in the frequency of stuttering on that word in all but sentence-final position. In nonfinal positions, the frequency of stutterers averaged 53% for focused words and 25% for nonfocused words. While previous results have indicated that stuttering is generally more likely on the first few words of a sentence than on later words, we found that the number of stuttering episodes decreased on successive words only if the later words in the sentence do not receive contrastive stress. The influence of sentence focus on $F0$ peaks throughout the course of the test sentences was found to be essentially the same as has previously been reported for normal speakers. The effects of contrastive stress on the $F0$ patterns were similar to those found for stuttering frequency, suggesting the possibility of a common underlying factor.

S16. Acoustic analysis of amyotrophic lateral sclerosis speech. Anthony J. Caruso, Kim A. Wilcox, J. Anthony Seikel, and Patti Haight (Department of Speech-Language-Hearing: Sciences and Disorders, 290 Haworth Hall, University of Kansas, Lawrence, KS 66045)

Seventeen victims of amyotrophic lateral sclerosis (ALS), were recorded as they produced /p t k b d g/ in a CVC syllable embedded in a carrier phrase. The duration of the stop consonant closure period, the release burst, and the aspirated phase of each of the stops, as well as the total duration of all pre- and post-stop vowels was determined spectrographically. In general, the ALS speakers produced highly varied speech which included numerous instances of voicing throughout voiceless segments, frication of voiced and voiceless stops, and multiple release bursts.

Comparisons of group mean durations, to those of normal speakers, indicates that the effects of decreased speaking rate in ALS are not evenly distributed across segments or portions of segments. Instead, some segment durations in ALS speech remain within normal limits while others increase relative to those seen in normal speech. Thus, the most sensitive acoustic/articulatory measure of the speech deterioration associated with ALS is achieved by comparing two or more durational measures with speakers. [Work supported by Kansas City Regional ALS Research Center, and NIH.]

S17. Using speech as an index of alcohol intoxication. Chris S. Martin and Moshe Yuchtman (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Although laypersons and low-enforcement officers routinely use perceived speech quality as an index of alcohol intoxication, there has been little research on the accuracy and limitations of this ability. In a recent study, we found that listeners could reliably discriminate between sentences produced by talkers speaking under sober and intoxicated conditions and that experienced listeners performed significantly better than naive listeners [Pisoni, Hathaway, and Yuchtman, SRL-PR #11, 109–172 (1985)]. In the present study, we compared the performance of college students and law-enforcement officers who were instructed to determine whether recorded sentences were produced under intoxication. In addition, digital signal processing techniques were used in an attempt to define changes in the acoustic-phonetic structure of speech under intoxication. The implications of this study for the role of speech as a reliable index of intoxication will be discussed. [Work supported by General Motors Research Laboratories.]

S18. Intra-speaker analysis of the nasal consonant [m]. H. J. Oyer, Y. Qi, C. Lambert, and B. Crowe (Speech and Hearing Science Section, Room 325 Derby, The Ohio State University, Columbus, OH 43210)

The acoustic spectrum of the nasal consonant [m] was analyzed from utterances of subjects as they produced it (1) in monosyllables, (2) in the reading of a paragraph, and (3) in attempts to disguise their voices while speaking a sentence. Findings indicate that although there is considerable intra-subject variability in the overall spectrum of the [m] consonant, the first formant remains remarkably constant. LPC analysis was employed in extracting acoustic characteristics. The results of this study have implications for speaker identification.

S19. Age-related differences in speech intensity among females. Richard J. Morris and W. S. Brown, Jr. (IASCP-ASB 63, University of Florida, Gainesville, FL 32611)

Among the physiologic consequences of aging is a reduction of vital capacity. Since the volume of air available may influence speech output, studies of age-related speech differences have included intensity as a variable. These studies have produced conflicting results, some experimenters have reported that intensity range and maximum intensity are reduced as a function of age, while others reported increased mean conversational intensity associated with aged speech. The purpose of this study was to compare the intensity range, maximum intensity, and mean conversational intensity between a population of geriatric women and a population of younger women. Two groups of 25 women—one 20–30 years old and another 75 years and over—who had no history of speech problems and exhibited normal hearing (for age level) served as subjects. They performed three repetitions of /ɑ/ for 5 s each at a typical, maximum, and minimum intensity level. Recordings of these productions were digitized and analyzed for mean intensity, intensity mode, and intensity distribution. Results will be discussed in relation to a general model of aging.

S20. The differential performance of older speakers in initiating and terminating phonation as a function of health status. Michael D. Trudeau, Dawn Mosca, and Herbert J. Oyer (Speech and Hearing Science Section, The Ohio State University, 154 North Oval Mall, Columbus, OH 43210)

The purpose of this study was to determine the utility of vowel initiation tune (VIT) and vowel termination time (VTT) in discrimination between healthy and unhealthy elderly speakers. Sixty-three adults age 60–94 years sustained /a, e, i, o, u/ for at least 1 s. From audio recordings of these productions means and standard deviations of VIT and VTT were derived. A MANOVA based on speaker gender (36 females, 27 males) and speaker health status (42 unhealthy, 21 healthy) yielded a significant difference in performance in VIT as a function of health status but for gender on the interaction. A subsequent linear discriminant analysis using mean VIT correctly classified subjects in the unhealthy group with 33.3% accuracy (3 times greater than chance levels) and subjects in the healthy group with 95.2% accuracy. These results support the hypothesis that the degree of coordination between respiration and phonation reflects the health status of the older adult.

S21. Age-related differences in phoneme durations among females. W. S. Brown, Jr. and Richard J. Morris (IASCP–ASB 63, University of Florida, Gainesville, FL 32611)

Reduced speech rate has been consistently reported for geriatric speakers when compared to the speech of younger adults. Only recently have the specific aspects of the speech signal associated with reduced rate been identified. The purpose of this study was to examine vowel and consonant durations using certain physiologic measures to determine their effect on age-related differences in speech rate. Two groups of 25 women—one 20–30 years old and another 75 years and older—who had no history of speech disorders and exhibited normal hearing (for age level) served as subjects. They repeated a series of CV, VCV, and VC syllables using the consonants /p/, /b/, /t/, /d/, /s/, and /z/ combined with the vowel /ɑ/, which were spoken in the carrier phrase "Speak 3# again." Intraoral air pressure and the voicing signal were recorded to provide vowel and consonant temporal information. Consonant duration did not differ significantly between the two age groups. Conversely, vowel durations were greater for the older females. These results are in agreement with previous studies and will be discussed in relation to a general model of aging.

S22. The effects of orthognathic osteotomy on formant structure. Jeanne George and Michael D. Trudeau (Speech and Hearing Science Section, The Ohio State University, 154 North Oval Mall, Columbus, OH 43210)

The objectives of this study were to determine the effect of orthognathic surgery on vowel production and the persistence of this effect beyond the immediate post-surgery period. Seven females provided cephalometric radiographs pre- and post-surgery and speech samples of /hd/ syllables for acoustical analysis 1 to 2 days prior to surgery, immediately following removal of intermaxillary fixation, and again 3 to 4 weeks later. Spectrographic analysis of the vowels' first two formants revealed little change between any of the three intervals. The only exception was a significant inverse relationship between vertical alteration of the oral cavity and change in $F2$ for front vowels. This relationship appeared only in comparing formant change between the presurgery sample and the final sample obtained 3 to 4 weeks after intermaxillary fixation had been removed ($r = -0.865$, d.f. $= 5$).

S23. Analysis for synthesis of arabic pharyngealized sound. A. Rajouani, M. Najim, A. Mouradi, and D. Chiadmi (L. E. E. S. A., Faculte des Sciences B. P. 1014, Rabat, Morocco)

Our interest in extracting relevant acoustic parameters and formulating phonological rules for a project oriented towards realization of text-to-speech synthesis by rule system for the Modern Standard Arabic (MSA) language. The present work includes the parametrization of the MSA pharyngealized consonants /ṭ/, /ṣ/, /ḍ/, /ð̣/ according a Klatt serial/

parallel formant synthesizer. Emphasis is made on the phonological level to define an adequate vowel system for the MSA by adding the pharyngealized vowels and to establish a set of rules dealing with the propagation of the pharyngealization feature. Evaluation of the synthesis strategy attests that the transition rules are independent of the presence of the pharyngealization. Synthetic speech generated from the developed synthesis-by-rule program is validated by Moroccan native informants. [Work partially supported by UNESCO.]

WEDNESDAY MORNING, 14 MAY 1986 — WHITE ROOM, 8:30 TO 10:20 A.M.

Session T. Underwater Acoustics III: Seabed Interaction

Peter H. Rogers, Chairman
School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332

Chairman's Introduction—8:30

Contributed Papers

8:35

T1. Effects of sediment parameters on data analysis of low-frequency beamformed bottom-propagating modes. G. E. Ioup (Physics Department, University of New Orleans, New Orleans, LA 70148), M. F. Werby (Code 221, NORDA, NSTL, MS 39529), and G. J. Tango (Code 223, NORDA, NSTL, MS 39529)

It has been previously shown [Tango, Werby, and Wooten, J. Acoust. Soc. Am. Suppl. 1 **78**, S71 (1985)] that shallow water sediments typically exhibit characteristic mode angular spectra when sampled by a towed or synthetic aperture array. In this paper, we further examine systematic effects of shallow layer thickness, compressional and shear velocity values, and depth distribution on simulated FFT-beamformed array responses. For a given source frequency and geobottom model, an increase in layer thickness simply increases the trapped mode response from the layer in question at the expense of leaky modes, and trapped modes from other depths. For sufficiently large numbers of layers and/or strong velocity gradients, clear critical angle information is complicated or lost in the beamformed pressure field response. Mode angular spectra are essentially independent of sediment shear rigidity for $v_S < v_{\text{water}}$; for larger shear velocity values, an additional attenuation as well as shear specific modes can result. Although effects of increased P and S wave sediment attenuation will effect observed mode amplitude [Frisk and Lynch, J. Acoust. Soc. Am. **76**, 205–216 (1984)], these effects are generally below the side lobe noise and resolution of the present beamforming algorithm. By using an alternative mode sampling strategy, as well as additional beamformed signal attributes, it is possible that improved accuracy and resolution can be obtained.

8:50

T2. Full-wave modeling of comparative deep sea bottom-interacting propagation to ocean bottom and sub-bottom receivers. G. J. Tango (Code 223, NORDA, NSTL, MS 39529), R. P. Wooten (ODSI, 6110 Executive Boulevard, Rockville, MD 20850), and M. F. Werby (Code 221, NORDA, NSTL, MS 39529)

A full-wave FFP reflectivity method [Schmidt and Tango, Geophys. J. R. Astron. Soc. (December 1985)] is applied to the problem of determining the characteristics of long-range, low-frequency signal propagation levels, at and beneath the seafloor, as seen by comparative horizontal and vertical arrays of hydrophones and geophones. Seisacoustically equivalent deep sea models for real variations in both fine scale shallow sediment and gross deep crustal velocity/depth structure are examined, to determine their effect on predicted signal level versus range and depth. Synthetic cw transmission loss and broadband horizontal and vertical seismic profile data reveal an overall trend of monotonically decreasing signal level with depth, interrupted by localized zones of relatively lower signal loss (3–7 dB), in agreement with synthetic and experimental borehole data for exploration seismic scenarios [Ganley and Kanasewich, J. Geophys. Res. (1981)]. Environmental and experimental factors controlling signal strength versus depth behavior are presented, and problems for further study are outlined.

9:05

T3. Effect of nonlinear frequency dependence of seabottom sound attenuation on low-frequency acoustic response in shallow water. Ji-xun Zhou[a] and Peter Rogers (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

In recent years several papers have been written on the existence and nature of the so-called "optimum frequency" (F_{opt}) for acoustic propagation in shallow water. These papers characterize the effects of various basic physical and environmental parameters that influence F_{opt} (such as water depth, sound velocity profile, sediment type, the depth and separation of the source–receiver pair, sediment sound-speed gradient, shear wave couple loss, etc.), but do not consider the frequency and the depth dependence of seabottom sound attenuation. In this paper, some experimental data on low-frequency response (30–2000 Hz) in shallow water are presented. The interesting result is that over the frequency range where F_{opt} should occur according to the existing theories, no apparent optimum frequency is observed. In order to explain these experimental results, this paper uses a normal mode computer program to examine the frequency dependence of sound propagation for various canonical models of seabottom structure in shallow-water regions (including thin top surface transition layer with strong positive sound-speed gradient of 1000 s^{-1} and negative attenuation gradient) allowing for the possibility that attenuation in the sediments may have a nonlinear frequency dependence. The results show that the experimental low-frequency responses, including the disappearance of the optimum frequency, are consistent with the earlier results of J. X. Zhou [J. Acoust. Soc. Am. **78**, 1003–1009 (1985)], i.e., that bottom attenuation in the low-frequency range has frequency power law of f^n, with values of n ranging from 1.5 to 2.0, although the results could also be explained to some degree by a seabed top transition layer. [a] Ji-xun Zhou on leave (1985) from the Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China.

9:20

T4. An investigation of the measurement sensitivities of acoustic remote profiling to bottom type and frequency. R. S. Bailey, T. L. Henderson, J. W. Maxwell, and E. A. Tschoepe (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

A method for remote geometric profiling of rough interfaces, which makes use of angular information obtained with a broadband monopulse sonar operating at low grazing angles, has been previously demonstrated in the analysis of backscattering data taken in lake and ocean environments [J. W. Maxwell and T. L. Henderson, J. Acoust. Soc. Am. Suppl. 1 78, S72 (1985)]. Since this method is sensitive to multiple scattering from different scattering centers and interference such as glint, sound penetration will produce false variations in the estimated profile. Analysis of data has often revealed such variations to exceed predicted values, frequently exaggerating the bottom irregularities to a marked degree. A remote profiling system was developed to conduct laboratory experiments to simulate remote profiling of the ocean bottom. Water tank experiments were conducted at selected frequencies from 30 to 100 kHz and pulse widths from 0.1 to 1.0 ms. The bottom types simulated consist of a sandy bottom, gravel over sand, gravel over a hard bottom, and a rough hard bottom. The variation of the profiles was analyzed with respect to the presumed degree of penetration of the sound into the bottom. As expected, the profiles were markedly smoother when frequency diversity was exploited.

9:35

T5. *In-situ* measurements of selected geo-acoustic properties of carbonate sediments at the Great Bahama Bank. Richard H. Bennett (Naval Ocean Research and Development Activity, Code 360, NSTL, MS 39529) and Tokuo Yamamoto (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

In-situ measurements were made of geo-acoustic properties of the various marine sediments at the Great Bahama Bank in November 1985. The depth profiles of density and porosity of the sediment were measured by an electrical conductivity probe. The depth profile of permeability was measured by an *in-situ* parameter. The water-wave-induced pressure and ground motion were measured by a bottom shear modulus profiler. Continuous cores of sediments were also taken for laboratory analyses. This paper presents the experimental techniques and the measurements. Two separate papers in this meeting present measurements of the complex velocities of compressional and shear waves and propagation of acoustic normal modes using *in-situ* and laboratory measurements from this field test. An ELAC sub-bottom profiler was used to survey the carbonate sediments to determine lateral and vertical variability. [The work was sponsored by ONR Code A25GG and Code 425UA.]

9:50

T6. Propagator matrix method for calculation of normal modes in a stratified ocean overlying inhomogeneous anisotropic porous beds. Mohsen Badiey (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149)

A practical algorithm for the calculation of acoustic normal modes at excitation frequencies of 50 to 5000 Hz in a shallow stratified ocean overlying a transverse isotropic poroelastic sediment bed is developed. The Biot–Willis stiffness matrix which describes poroelastic anisotropy in terms of physical properties of sediments is used to model the bed. The propagator matrix method is used to solve the differential equations for the motion stress vectors in both layered sediment and water. The method of impedance matching is used to obtain the eigenvalues of the system numerically. Using measured *in-situ* properties of sediment (presented separately in this meeting), the effect of sediment anisotropy and inhomogeniety on acoustic wave propagation is studied for shallow water propagation at the Great Bahama Bank. [The work was sponsored by ONR Codes 425UA.]

10:05

T7. Laboratory measurements of selected geo-acoustic properties of carbonate sediments at the Great Bahama Bank. Tokuo Yamamoto (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149) and Richard H. Bennett (Naval Ocean Research and Development Activity, Code 360, NSTL, MS 39529)

The complex shear modulus of various carbonate sediments taken from the Great Bahama Bank was measured mechanically at very low shear strain amplitudes of the order of 10^{-7} using a high-precision torsional resonant column apparatus. Permeability and porosity of the sediments were also measured. The measurements were made under simulated *in-situ* stress conditions. The data from the laboratory measurements were compared with those from the *in-situ* measurements which are presented separately in this meeting. Finally, the depth profiles of the velocities and the attenuation of the compressional waves and the shear waves were delivered as a function of frequency ranging from 50–5000 Hz. Our velocity and attenuation profiles were compared with the profiles recommended by E. Hamilton. [The work was sponsored by ONR Codes 425UA and 425GG.]

Plenary Session

Floyd Dunn, Chairman
President, Acoustical Society of America

Presentation of Awards

Presentation of Gold Medal to James L. Flanagan
Presentation of Biennial Award to William E. Cooper

Philip M. Morse: Reminiscences

K. Uno Ingard
Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

WEDNESDAY AFTERNOON, 14 MAY 1986 ROCKEFELLER ROOM, 2:00 P.M.

Meeting of Accredited Standards Committee S2: Mechanical Shock and Vibration

to be held jointly with the

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

P. H. Maedel, Jr., Chairman S2
Westinghouse Electric Corporation, Lester Branch, P. O. Box 9175, Lester, Pennsylvania 19113

G. Booth, Chairman, Technical Advisory Group for ISO/TC 108
220 Clark Avenue, Branford, Connecticut 06405

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

Session U. Noise IV: Future Directions in Acoustics of the Workplace

John Erdreich, Chairman
National Institute for Occupational Safety & Health, Cincinnati, Ohio 45226

Chairman's Introduction—1:30

Invited Papers

1:35

U1. Economic trends in the U. S. industry. Ron E. Kutscher (Office of Economic Growth and Employment Projections, Bureau of Labor Statistics, Washington, DC 20212)

Is the U.S. losing its industrial base? This paper presents evidence that the industrial sector as a whole is in healthy shape, but a small number of manufacturing industries are in deep trouble. Our findings indicate that the well-reported shift to service industries is not really evidence of a declining industrial base or "de-industrialization," but rather part of a long-term structural change in the U.S. economy. That structural change may have accelerated in the last business cycle trough but goods producing industries including manufacturing have bounced back during the recent upswing and have reached a new peak.

2:05

U2. Clinical, medical, and behavioral research needs to foster improved hearing conservation. P. W. Alberti (Department of Otolaryngology, University of Toronto, and Mount Sinai Hospital, 600 University Avenue, Toronto, Ontario M5G 1X5, Canada)

Outstanding medical problems include the identification of noise susceptible individuals; development of new medical management for acute noise exposure, the relationship of tinnitus to noise type and level, and continuing study of the relationship between noise and other diseases; detailed studies of the interaction of peripheral and central lesions, particularly the central aging effects, is also required. What is the relationship of noise exposure and hearing loss on speech? Definition of appropriate presbycusis corrections are required which are related to cutoff levels for compensation. The effect of differing noise exposures and, in particular, intermittency of exposure is not clear. What is the effect of currently popular eccentric work weeks with 3×12-h shifts and 4 days off? Is there need to control overtime from total exposure in noise industries? How important is sociocusis? What are the long-term outcomes of the hearing conservation programs? What is the safety risk associated with hearing protectors? What stratagem of hearing conservation should be used on an already hearing-impaired work force? The problems of monitoring hearing protectors and the need to service hearing protectors regularly will be stressed.

2:35

U3. Research needs for standards development regarding noise in the workplace. Kenneth McK. Eldred (Ken Eldred Engineering, P.O. Box 1037, Concord, MA 01742)

The methodology for measurement and evaluation of noise in the workplace has evolved from the work of many organizations in the USA and abroad. In the 1950s the Department of Defense initiated major hearing conservation programs in the services. In the late 1960s the Department of Labor promulgated its first noise standards under the Walsh Healy Act, and in the early 1970s other agencies including the Environmental Protection Agency, Department of Transportation, and the Bureau of Mines initiated programs in these areas of responsibility. Also, during the 1970s the national and international voluntary standards sectors began the development of a set of standards for measurement and evaluation of noise in the workplace. In the initial planning of these standards it became apparent that research would be needed to develop meaningful standards, particularly in areas outside DOD. This paper outlines the current and proposed future plans for voluntary standards in this area and identifies a set of research tasks and objectives that are required to support a comprehensive standards development effort.

Introduction for Panel Discussion on Future Directions in Acoustics of the Workplace

Recently, there have been expressions of concern that the acoustics community has not been properly cognizant of changing trends in American industry and how they may affect the role of the acoustician in the workplace. The purpose of this panel discussion is to raise issues of noise, hearing, and regulation for discussion of the changes in direction, if any, which should be considered. Four individuals have been invited as formal discussants of the presented papers. Audience participation is encouraged to add other perspectives.

Invited Discussants:

Josef Miller, *Kresge Hearing Research Institute, Ann Arbor, MI*
Eula Bingham, *University of Cincinnati, Cincinnati, OH*
Allen Cudworth, *Liberty Mutual Insurance, Hopkinton, MA*
Walter Haag, *National Institute for Occupational Safety and Health, Cincinnati, OH*

Open Discussion 4:05–4:35

WEDNESDAY AFTERNOON, 14 MAY 1986 SAVOY ROOM 1:30 TO 5:15 P.M.

Session V. Physical Acoustics V: Ultrasound in Medicine

Michael E. Haran, Chairman
D. H. R., Inc., 6849 Old Dominion Drive, McLean, Virginia 22101

Invited Papers

1:30

V1. Ultrasonic tissue characterization: A Review. Joie Pierce Jones (Department of Radiological Sciences, University of California Irvine, Irvine, CA 92717)

Present medical ultrasound systems are based on envelope detection methods and therefore display only echo intensity information. However, phase information is also recorded by the transducer which is a pressure-sensitive device, but is not utilized in commercial display or measurement schemes. This additional information may be of diagnostic significance since the interaction between sound and tissue acoustical properties can be correlated with specific pathological states. Thus, in principle, *in vivo* techniques could be devised which would extract and separate the medically significant features of the ultrasound interactions with tissue and would display ultrasonic tissue signatures appropriate for a differential diagnosis. The development of such quantitative techniques for the measurement of ultrasonic tissue parameters and/or the display of ultrasonic tissue signatures has become known as ultrasonic tissue characterization. In this paper we review the physical basis for ultrasonic tissue characterization, describe several characterization schemes currently being explored, and attempt to assess likely future developments in the field as well as the importance such techniques may play in the future of diagnostic medicine.

2:00

V2. Diagnostic imaging of the nonlinear acoustic parameter B/A. Peter A. Lewin (Department of Electrical and Computer Engineering and Biomedical Engineering and Science Institute, Drexel University, Philadelphia, PA 19104)

One of the most promising applications of ultrasound in medicine, currently in the experimental trial stage, is imaging based on the nonlinear propagation parameter B/A. Evidence is currently emerging that B/A is related to the structural characteristics of tissue and information on B/A variations in tissue may provide a new avenue in diagnostic ultrasound tissue characterization. This presentation will briefly outline the theoretical background and operation principles of real-time nonlinear parameter imaging systems, including both pulsed and cw devices. Subsequently, the differences between conventional, real-time B-scan imaging and the B/A

imaging will be presented. Different nonlinear imaging algorithms, including those for the more desirable pulse-echo mode of operation, will be discussed and the fundamental limitations of this novel imaging facility will be reviewed. Possible modifications in the currently used B/A imaging procedures to improve B/A picture quality will also be discussed, and the potential applications of B/A imaging in ultrasound tissue characterization will be pointed out.

2:30

V3. Ultrasonic assessment of burn injury. John H. Cantrell and William T. Yost (NASA–Langley Research Center, Mail Stop 231, Hampton, VA 23665-5225)

The quantitative assessment of burn depth for victims of severe thermal injury is one of the most challenging problems facing the burn surgeon today. We have developed a high-resolution, ultrasonic system that is capable of measuring the depth of skin burns that agrees to within 5% of the values obtained independently from histological sectioning. Such accuracy is well within surgical requirements. Ultrasonic measurements of the time–temperature dependence of cutaneous burn depth together with a theoretical model of the burning process reveals collagen denaturation to be the progenitor of thermal necrosis. This finding provides an objective basis upon which to diagnose burn injury and establishes the ultrasonic technique as a viable alternative to subjective diagnostic methodologies currently in use.

Contributed Papers

3:00

V4. The cause of the focus artifact in clinical sonography. Paul S. LaFollette, Jr., and Marvin C. Ziskin (Departments of Diagnostic Imaging and Computer and Information Sciences, Temple University, Philadelphia, PA 19140)

The focus artifact is commonly observed spurious echo enhancement that is seen as a linear band in clinical sonograms. It occurs at a distance from the transducer equaling the focal length, and although believed to be related to a focusing effect of the sound beam, its production has never been adequately explained. The notion that focusing concentrates the energy producing a more intense reflection in the focal region, although correct, is an inadequate explanation because the number of reflectors irradiated is reduced proportionally to the intensity increase to those receiving the ultrasound. If this was all there is to the focusing effect, the transducer would record the same amount of sound in either the focused or unfocused case. In order to provide an adequate explanation, a mathematical model has been developed that predicts the geometric and intensity distortions that occur in a sonogram produced by a clinical instrument when the sound beam passes through a converging lens. By considering a focusing transducer as a plane transducer with a converging lens placed in contact with its surface, this model does show a marked relative echo enhancement occurring at a distance from the lens equaling the focal length. In addition to the theoretical analysis and prediction, this artifactual echo enhancement has been demonstrated in several test objects in the laboratory and also in clinical sonograms.

3:15

V5. Medical flow imaging with ultrasound. Richard K. Johnson (Quantum Medical Systems, 1065 12th Avenue, NW, Issaquah, WA 98027)

Recent technology advances have made possible commercial machines which produce real-time images of blood flow for medical applications. Design approaches will be discussed for the issues of velocity range and resolution, tissue and flow differentiation, two-component image processing, and frame rate. A videotape of clinical flow images will be shown.

3:30

V6. Ultrasonic assessment of skin and wound with the scanning laser acoustic microscope. D. L. Steiger (Bioacoustics Research Lab, University of Illinois, 1406 W. Green Street, Urbana, IL 61801), J. E. Olerud (Department of Medicine, University of Washington, Seattle, WA 98195), M. A. Riederer-Henderson (Veterans Administration Medical Center, Seattle, WA 98108), and W. D. O'Brian, Jr. (Bioacoustics Research Lab, University of Illinois, Urbana, IL 61801)

The ultrasonic attenuation coefficient and speed of canine wound tissue and adjacent skin have been studied at wound ages between 7 and 50 days with the scanning laser acoustic microscope at 100 MHz. Water, total collagen, and acid soluble collagen contents were determined for both wound and skin. Attenuation coefficient and speed were consistently greater for the skin than for the wound. Water content was generally higher and total collagen was generally lower in the wound tissue. Total collagen increased in the wound and its difference between wound and skin decreased with maturation. Attenuation coefficient (A in dB/mm) and speed (c in m/s) were highly significant with both total collagen and water. Mathematically, via least-squares analysis, $A = 1.67C + 6.3$, $A = -1.77W + 154$, $c = 6.5C + 1460$, and $c = -5.2W + 1923$, where C is the total collagen (as percentage on a wet weight basis) and W is the percent water. [This work supported by NIH AM 21557 and CA 36029.]

3:45

V7. Measurement of low-intensity highly focused ultrasound. George H. Myers (Medsys, Inc., 201 Route 17, Rutherford, NJ 07070), Peter A. Lewin, and Mark E. Schafer (Department of Electrical and Computer Engineering and Biomedical Engineering and Science Institute, Drexel University, Philadelphia, PA 19104)

The requirements for precise determination of acoustic output for different types of medical diagnostic equipment have made it important to develop methods of measuring ultrasonic field parameters at high megahertz range of frequencies. More specifically, of particular interest here is the measurement of pulsed beams of small focal diameters (of the order of a wavelength), generated by the low total power output devices. These low power levels are in general on the order of 50 μW, which is below the sensitivity of current acoustic power measurements methods including radiation force balances and calorimeters. While the miniature wideband polymer hydrophones have the requisite voltage sensitivity and bandwidth to correctly record the pressure waveforms, their finite dimensions, typically 0.5 to 1.0 mm in diameter, lead to spatial averaging effects. To overcome this problem, a technique has been developed, and is now being tested which permits hydrophones to be used in these applications. The measurement procedure is based on the fact that: (a) the entire beam is intercepted by the hydrophone and, (b) the beam profile can be accurately determined by pulse-echo techniques. The pressure waveform is recorded from a calibrated hydrophone of known effective area, located at the focus of the beam. The total force on the hydrophone is determined by numerically integrating the beam profile. The peak pressure, intensity, and total transmitted power can then be directly determined. Experimental results will be presented and both the advantages and limitations of the proposed measurement procedure will be discussed.

4:00

V8. The use of frequency weighting in multifrequency holography. T. J. Teo and J. M. Reid (Department of Biomedical Engineering, Drexel University, Philadelphia, PA 19104)

Backpropagating an acoustical field that is measured on a planar surface using angular spectrum decomposition [J. Goodman, *Introduction to Fourier Optics* (McGraw-Hill, New York, 1968)] can be used to reconstruct a source region or a scattering region. The reconstruction algorithm is inherently monochromatic and hence the range resolution is typically worse than the lateral resolution. In diffraction tomography, the same backpropagation algorithm is used and the summation over views is used to improve the overall resolutions [A. Devaney, Ultrason. Imag. **4**, 336–350 (1982)]. In the case of *B*-scan imaging, the reverse is true. The use of short pulse provides a good range resolution while the lateral resolution is limited by the beamwidth. It is therefore advantageous to combine the wideband approach with the backpropagation. The use of multiple frequencies in backpropagating the acoustical field has shown to improve the range resolution [e.g., T. J. Teo and J. M. Reid, Acoust. Imag. **14**, 143–153 (1985)]. The two techniques can therefore be combined by suitably weighting the frequency distribution with the amplitude and phase spectra of a pulse. We will also show the effect of pulse shape on the image reconstructed and range resolution in particular.

4:15

V9. Application of the sinc basis moment method to the reconstruction of infinite circular cylinders. T. J. Cavicchi (Bioacoustics Research Lab, University of Illinois, 1406 W. Green Street, Urbana, IL 61801), S. A. Johnson (Department of Bioengineering, University of Utah, Salt Lake City, UT 84112), and W. D. O'Brian, Jr. (Bioacoustics Research Lab, University of Illinois, Urbana, IL 61801)

A solution to the ultrasonic scattering and inverse scattering problem has been obtained by solving the exact inhomogeneous Helmholtz wave equation via the sinc basis moment method. In this numerical study, the algorithm of Johnson and Tracy [Ultrason. Imag. **5**, 361–375 (1983)] has been applied to the reconstruction of an infinite circular cylinder which is subject to an incident cylindrical ultrasonic wave and is surrounded by a homogeneous coupling medium. For weak scatterers, successful reconstructions have been obtained using the known exact solution for the scattered field in the case of a circular object function as the input data for the algorithm. Five percent noise does not affect performance, nor does transmitter and receiver distance to the object for a lossless coupling medium. Also for weak scatterers and exact field data, the reconstruction quality improves for cylinders when the grid size increases because the discretized shape becomes more nearly circular. Use of either the discrete scattered field equations or the exact Bessel function series solution yielded identical results, verifying the validity of the sinc function expansions. [This work is supported by CA 09067.]

4:30

V10. Acoustic study of artificial blood substitutes. M. A. Barrett Gultepe, M. E. Gultepe, J. L. McCarthy, and E. Yeager (Ultrasonic Research Laboratory, Department of Chemistry, Case Western Reserve University, University Circle, Cleveland, OH 44106)

Absorption and velocity of sound were measured in various submicron size perfluorochemical (PFC) emulsions in water of the types considered for use as artificial blood substitutes. The attenuation of sound was measured in the frequency range 1 to 95 MHz with computer-assisted VHF and UHF send–receive apparatus. Concentrated aqueous emulsions of F-decalin, F-tributylamine, and F-phenanthrene stabilized by various surfactants including fluorinated surfactants were studied as well as Fluosol-43 emulsion, a commercially produced emulsion for use as an artificial blood substitute. The attenuation of sound waves in the abovementioned PFC emulsions can be theoretically described by heat conduction losses, viscous drag losses, and scattering of sound. In contrast to the majority of hydrocarbon/water emulsions, in PFC emulsions the contribution to acoustic absorption of the viscous drag losses and heat conduction losses are approximately equal to magnitude, though shifted in frequency. This is a result of the high density of perfluorochemicals. [Work partially supported by Advanced Biosystems Inc.]

4:45

V11. Three-dimensional model for piezoelectric ceramic mode vibration determination. M. Brissaud, L. Eyraud, and H. Kleimann (Department of Electrical Engineering and Ferroelectricity, I.N.S.A., 69621 Villeurbanne Cedex, France)

The demand for imaging properties of ever increasing quality in ultrasonic echography requires an analytical tool able to give easy predictions of the transducers' performances. Though algorithms based on the finite element method are able to solve the steady-state problem [O. C. Zienkiewicz, *The Finite Element Method* (McGraw-Hill, New York 1977)], the analytical methods are often preferred because the numerical approaches do not give sufficient insight into the physical parameters which should be kept under control in the design of piezoelectric transducers. The aim of this paper is the analysis of the vibration of rectangular or beam-shape ceramics transducers used in acoustical imaging. Coupling between thickness and lateral modes is taken into account. The general electrical impedance of the transducer is

$$Z=\frac{1}{jC_0\omega}\left[1-\frac{h_{31}c_{33}^D-h_{33}c_{13}^D}{(c_{11}^D+c_{12}^D)\,c_{33}^D-2c_{13}^D}\right.$$
$$\times\frac{h_{31}}{\beta_{33}^S}\left(\frac{1}{\cos(\omega a_1/2v)}+\frac{1}{\cos(\omega a_2/2v)}\right)$$
$$\left.-k_t^2\frac{1-2h_{31}c_{13}^D/h_{33}(c_{11}^D+c_{12}^D)}{1-2c_{13}^{D2}/(c_{11}^D\;c_{12}^D)\,c_{33}^D}\,\frac{\tan(\omega a_3/2v_3)}{\omega a_3/2v_3}\right],$$

where $a_i, c_{ij}^D, h_{ij}, C_0$ are the geometrical dimensions, the stiffness and piezoelectric constants, and the capacitance of the ceramic, respectively. When h_{31} and c_{13}^D vanish, the electrical impedance formula becomes that given by W. P. Mason [*Physical Acoustics* (Academic, New York, 1964), Vol. 1, part A].

5:00

V12. Nucleation of acoustic cavitation from a gas-filled crevice: Anthony A. Atchley[a] (Center for Ultrasonics and Sonics, Yale University, P. O. Box 2159 Yale Station, New Haven, CT 06520) and Andrea Prosperetti (Department of Mechanical Engineering, 127 Latrobe Hall, The John Hopkins University, Baltimore, MD 21218)

The nature of the cavitation nucleus, although the subject of considerable attention by physical acousticians in the past, continues to be important, in part because of the recent increased investigation of cavitation in biological systems. At a previous meeting Atchley and Crum [J. Acoust. Soc. Am. Suppl. 1 **76**, S64 (1984)] presented some preliminary results of a rederivation of the crevice model of cavitation nucleation. In the present paper, the final results of this work are discussed. According to the model, a gas pocket stabilized at the bottom of a conical crevice residing on a hydrophobic solid impurity acts as the cavitation nucleus. In order for cavitation to occur, the nucleus must be mechanically unstable and the liquid–gas interface must reach the receding contact angle. The theoretical predictions of this model agree quite well with the results of measurements made in water. [Work supported by ONR and NSF. AAA would like to acknowledge the support of the Hunt Fellowship.] [a] On leave from Dept. of Physics, Naval Postgraduate School, Monterey, CA 93943.

Session W. Physiological Acoustics IV and Psychological Acoustics V: Effects of Noise and Other Maskers

Brian R. Shelton, Chairman
Bell Northern Research Department 2Z33, Box 3511 Station C, Ottawa, Ontario K1Y 4M7, Canada

Contributed Papers

1:15

W1. Evaluation by computer simulation of stopping rules for audiological ascending test procedures. Lynne Marshall and Thomas E. Hanna (NSMRL, Box 900, SUBASE NLON, Groton, CT 06349)

Stopping rules for ascending audiological test procedures were evaluated by Monte Carlo simulation. The stopping rules differed in the minimum number of correct responses required at a level, whether these responses occurred on half or a majority of the ascents, whether a limit was placed on the number of ascents, and whether all or only the most recent ascents were considered. Simulated threshold values ranged from − 12 to 92 dB SPL in steps of 1 dB. Slopes of the psychometric functions ranged from 0.1 to 1.0 in steps of 0.1. For each procedure, 200 threshold determinations were simulated for each combination of slope and threshold value. For all procedures, shallow slopes resulted in thresholds closer to the level giving 50% detection than did higher slopes, which were roughly 2.5 dB above 50% on the psychometric function. Shallow slopes also resulted in decreased consistency across threshold measurements, an increased number of trials required for threshold estimates, and a higher proportion of estimates that had to be repeated to obtain threshold. Stopping rules using a two-response criterion were faster than those using a three-response criterion, with only a small decrease in consistency. Among stopping rules using the same number of responses for criterion, differences were seen primarily in efficiency at shallow slopes, particularly for procedures using a three-response rather than a two-response criterion for stopping.

1:30

W2. A method to estimate fluctuations in criterion and sensitivity. Thomas E. Hanna (Auditory and Communication Sciences Department, Naval Submarine Medical Research Laboratory, Box 900, SUBASE-NLON, Groton CT 06349)

The theory of signal detection (TSD) typically assumes variable sensory information and a fixed criterion. A fixed criterion is assumed because of our inability to distinguish sensory variance from criterion variance based only on a summary of hit, false alarm, miss, and correct rejection percentages. The present paper develops a model which distinguishes sources of variance by considering two-trial response patterns. The model assumes that criterion and sensitivity are normally distributed over the course of an experiment but are relatively constant over successive trials, i.e., they vary slowly. By computing statistics over pairs of successive trials it is possible to estimate the proportion of variance that is attributable to fluctuations in criterion and that is attributable to fluctuations in sensitivity.

1:45

W3. The effect of masker bandwidth on signal detectability. Richard S. Bernstein (Department of Otolaryngology, Albert Einstein College of Medicine, Bronx, NY 10461) and David H. Raab (Department of Psychology, Brooklyn College, Brooklyn, NY 11210)

The effect of masker bandwidth on the detectability of narrow- and wideband signals was investigated. The signals were pure tones and noise bands. The noise bands varied from 50–792 Hz, center frequency being 1500 Hz. Stimuli were computer generated and the noise bands had nearly rectangular spectra. Log–log plots of threshold as a function of masker bandwidth (Fletcher plots) were found to be well fitted by two line segments. An iterative least-squares procedure was employed to determine slopes and intercepts of the line segments. These parameters were found to depend on the bandwidth of the signal. An energy detection model [Green, J. Acoust. Soc. Am. 32, 121–131 (1960)] was modified by the addition of internal variability that is bandwidth-dependent. This modified energy-detection model was able to predict the results obtained with narrow bandwidth signals. The relative magnitude of internal variability was estimated and the ratio of internal-to-external variability agreed with estimates presented by Raab and Goldberg [J. Acoust. Soc. Am. 57, 437–447 (1975)]. In contrast, the results obtained with wideband signals were not predicted by the simple model. They may reflect the operation of a mechanism for the enhancement of spectral contours.

2:00

W4. Uncertainty and response time in identifying nonspeech sounds. James A. Ballas, Martin J. Sliwinski, and John P. Harding, III (Department of Psychology, Georgetown University, Washington, DC 20057)

The identification of short duration (< 500 ms) sounds taken from sound effects records was studied to determine if measures of causal uncertainty would be related to identification time. Twenty-five sounds were randomly presented to listeners who were instructed to respond as soon as they had a reasonable guess about the cause of the sound. Identification times averaged across listeners ranged from 921 ms for a doorbell to 5982 ms for a monkey cry. All identifications for a sound were combined, sorted for similar responses, and used to compute the uncertainty metric from information theory. For the sounds used, the uncertainty metric was significantly correlated with the log of the average response time ($r = +0.59$). Correlations for individual subjects ranged from + 0.15 to + 0.71.

2:15

W5. The effect of amplitude and spectral uncertainty on masking produced by small numbers of sinusoids. Donna L. Neff and Brian P. Callaghan (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

Threshold for a 1000-Hz sinusoidal signal in the presence of multicomponent maskers was measured for four normal-hearing listeners. Masker and signal were 200 ms, presented simultaneously. Maskers were composed of 2 to 200 sinusoids whose frequencies were drawn at random from a range of 300–3000 Hz, excluding the signal frequency. Component frequencies, amplitudes, or both were either fixed or randomized across the two intervals of the 2AFC trials; new frequencies were drawn for each trial. For maskers with more than ten components, there was little effect of randomizing either component frequency or amplitude. For maskers with ten or fewer components, randomizing frequency across intervals produced large amounts of masking but there was little effect of randomizing amplitude. Similar results were obtained when components were eliminated from the critical band around the signal frequency. Under some conditions, listeners apparently cannot use a single-filter detection strategy even when it is advantageous to do so. [Work supported by NIH.]

2:30

W6. Detection of simple and complex tones in fixed and random conditions. Søren Buus, Edwin Schorer, Mary Florentine, and Eberhard Zwicker (Institute of Electroacoustics, Technical University of Munich, Munich, Federal Republic of Germany and Northeastern University, Boston, MA 02115)

Psychometric functions for detection of pure (220, 1100, or 3850 Hz) and complex tones in the presence of a 64 dB SPL uniformly masking noise were measured in a 21, 2AFC paradigm. The complex tone consisted of 18 equally intense components spaced about one critical band apart between 110 and 7260 Hz. In experiment 1, all levels for a single signal were presented in mixed order within a block of trials. Results for eight normal listeners show parallel psychometric functions for simple and complex tones. Thresholds are 43–44 dB SPL for the pure tones and about 37 dB SPL per tone for the complex tone. In experiment 2, all levels and signals were presented in random order within each block of trials. Results for four normal listeners show psychometric functions parallel to those in experiment 1. Thresholds are about 46 dB for the pure tones and 40.5 dB for the complex tone. These results are consistent with a multiband energy-detector model in which the decision is based on an unweighted sum of decision variables across an optimum selection of channels. [Supported by Deutsche Forschungsgemeinschaft and NIH-NINCDS R0 1NS 18280.]

2:45

W7. "Confusion effects" in simultaneous masking. John B. Mott[a] (Boys Town National Institute for Communication Disorders in Children, 555 N. 30th Street, Omaha, NE 68131) and Lawrence L. Feth (Department of Speech-Language-Hearing: Sciences and Disorders, University of Kansas, Lawrence, KS 66045)

A "confusion effect," the inability to discriminate the probe from the masker in certain stimulus configurations, has been well-documented in forward masking [e.g., D. Neff, J. Acoust. Soc. Am. **78**, 1966–1976 (1985)]. Similar effects are reported for simultaneous masking. Bimodal thresholds in simultaneous masking are observed when the masker envelope fluctuates and the probe duration is shorter than the mean interval between envelope peaks. The magnitude of the confusion effect, as measured by the difference between threshold estimates for a given masker and probe combination, is similar in simultaneous and forward masking. However, while confusion effects in forward masking are typically observed when the probe is near the masker frequency, bimodal thresholds in simultaneous masking are observed across a wider frequency region. This result does not favor a confusion hypothesis, since the spectral difference between masker and probe should reduce or eliminate confusion effects. [Work supported by NINCDS grant #T-32-NS07257.] [a] These data were collected as part of the dissertation research completed by J. B. Mott at the University of Kansas.

3:00–3:15

Break

3:15

W8. The significance of spectral synchrony in broadband signal detection. T. Houtgast (TNO Institute for Perception, Soesterberg, The Netherlands)

The masked threshold of a brief and broadband signal was determined for various patterns of temporal delays among the individual 1/3-octave-band contributions. The signal consisted of the sum of nine individual Gaussian-shaped tone pulses, such that (1) the effective bandwidth of each tone pulse was 1/3 octave, (2) the center frequencies were located at 1/3-octave intervals (from 500 up to 3170 Hz), and (3) the intensity ratios were such that all nine individual tone pulses had the same masked threshold (in pink noise) when presented individually. The masked threshold of the total stimulus was determined as a function of the temporal interrelation among the nine tone pulses. For the condition of perfect spectral synchrony (i.e., temporal coincidence of the peaks of the nine Gaussian-shaped envelopes), detection threshold was found to be about 6 dB better than in the case of a desynchronized stimulus for which the nine individual tone pulses were distributed randomly within a 100-ms interval (which is well within the traditional integration window for signal detection.) This phenomenon of synchrony-facilitated broadband detection has interesting implications, both for signal-detection theory and from a functional point of view.

3:30

W9. Human tone detection in continuous- and pulsed-noise maskers. David M. Pressel and Murray B. Sachs (Department of Biomedical Engineering, Johns Hopkins University, Baltimore, MD 21205)

A two-interval, forced-choice paradigm was used to measure the critical ratio of human subjects detecting a tone burst in continuous noise and noise gated simultaneously with the tone bursts. Noise spectrum levels of 35, 15, and -5 dB SPL/$\sqrt{\text{Hz}}$ were used; tone frequencies were 500, 1000, 2000, 4000, and 8000 Hz. At 4000 and 8000 Hz, critical ratios were significantly larger for the simultaneously gated noise than for continuous noise. There were no significant differences between the critical ratios for continuous and gated noise at frequencies of 500, 1000, and 2000 Hz. These results are consistent with those of Wier *et al.* [J. Acoust. Soc. Am. **61**, 1298–1300 (1977)] and extend those results to higher frequencies where differences between the continuous and simultaneously gated masking cases are greater.

3:45

W10. Effect of peripheral and central auditory lesions on auditory pattern perception. Frank E. Musiek and Marilyn Pinheiro (Dartmouth-Hitchcock Medical Center, Hanover, NH 03756)

Auditory patterns composed of three successive tone bursts of either $880^{(\text{Low})}$ or $1122^{(\text{High})}$ Hz were arranged to yield six types of patterns (i.e., LLH, LHL, HLL, HHL, HLH, LHH). The tones in the patterns had a 150-ms duration, 10-ms rise–fall and 200 ms interstimulus interval. A normal performance criteria for these monaurally presented patterns was established on 31 normal subjects. Pattern data were then collected on groups of subjects with documented lesions of the cochlea, brain stem, or auditory areas of the cerebrum. Results indicated that almost 90% of subjects with cochlear lesions ($N = 29$) performed within the normal criteria, while over 80% of the 29 subjects with cerebral lesions yielded abnormal results. Approximately one half of the 22 subjects with brain stem involvement demonstrated normal performance on the pattern task. An additional analysis revealed that a high percentage of subjects with lesions limited to one hemisphere demonstrated bilateral ear deficits for pattern identification.

4:00

W11. Temporary threshold shifts from attendance at a rock concert. William W. Clark and Barbara A. Bohne (Central Institute for the Deaf and Department of Otolaryngology, Washington University School of Medicine, St. Louis, MO 63110)

The relation between exposure level and hearing loss in rock concert attendees was studied. Six volunteer subjects, ages 16–44, participated. All except the 44-year-old had normal hearing sensitivity, as revealed by audiometric evaluations made immediately before the concert. They attended a Bruce Springsteen concert at the St. Louis Arena and returned to CID for another hearing test within 30 min following the concert. Noise exposure was assessed by having two subjects, seated at different locations in the arena, wear calibrated dosimeters during the event. Sixteen hours after the concert all subjects returned for a final audiometric evaluation. Results indicated that the average exposure level was 100–100.6 dBA during the 4½-h concert. Five of the six attendees had significant threshold shifts (< 50 dB) predominately in the 4-kHz region. Measures made 16 h after the concert and thereafter indicated that hearing returned to normal in all subjects. Although no PTS was observed, comparison of these data with studies of hearing loss and cochlear damage in animal models suggests that these subjects may have sustained some sensory cell loss from this exposure. [Work supported by NIOSH and NINCDS.]

4:15

W12. Behavioral and physiological measures of temporary threshold shift. Donald P. Gans (Speech Pathology and Audiology, Kent State University, Kent, OH 44242)

Physiological studies of temporary threshold shift (TTS) have produced conflicting evidence regarding the contribution of peripheral and neural components to the hearing loss. Under appropriate recording and analysis techniques, it was found that the cochlear microphonic and summating potential exhibit similar amounts of TTS as the whole nerve action potential (AP). An important assumption is that the AP is an accurate measure of hearing loss. Two independent studies have produced evidence that central auditory pathways are strongly affected during TTS which would invalidate the AP as an accurate measure. To examine this possibility, behavioral thresholds were determined in 14 gerbils prior to and following exposure to an 8000-Hz pure tone. Each gerbil was subsequently anesthetized, the middle ear surgically exposed, and an electrode inserted in the cochlea. AP post-exposure thresholds were compared to normative data for calculation of AP TTS. It was found that behavioral and AP TTS were not statistically different at any of the four test frequencies. These data are consistent with traditional theories of a cochlear origin for TTS. [Work supported by NSF.]

4:30

W13. Evoked response "forward-masking" functions in hearing-impaired chinchillas. Shalini Arehole, Richard J. Salvi, Samuel S. Saunders, and Donald Henderson (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

Previous research in psychophysics has shown that the time course of recovery from forward masking is prolonged in listeners with sensorineural hearing loss (Nelson and Turner, 1980). In order to evaluate the physiological basis for the change in the time course of forward masking, evoked response recordings were obtained from the inferior colliculus of the chinchilla both before and after noise induced hearing loss. Evoked response "forward-masking" functions were obtained by measuring the masker level required to produce a 50% reduction in the response to a probe tone (*re*: probe response alone) which followed the masker at various time intervals (2 to 100 ms). The time course of "forward masking" was estimated by fitting a time constant to the data. Animals were exposed to a pure tone that produced either 30–50 dB of asymptotic threshold shift or 25–30 dB of permanent threshold shift. The time constants fit to the "forward-masking" function were found to be prolonged in the region of hearing loss. In general there was a strong correlation between the time constant of recovery from "forward masking" and hearing loss. The implications of these results will be discussed. [Work supported by NIH 5-R01-NS16761-06 and NIOSH 1-R01-OH00364.]

4:45

W14. PET measurement of regional cerebral blood flow changes during noise stimulation. J. J. Sidtis, V. Dhawan, J. O. Jarden, S. C. Strother, and D. A. Rottenberg (Department of Neurology, Memorial Sloan–Kettering Cancer Center, 1275 York Avenue, New York, NY 10021 and Cornell Medical Center, New York, NY)

Positron emission tomographic (PET) studies of regional cerebral blood flow (rCBF) using the steady-state 0–15 carbon dioxide (CO_2) method allows within-subject stimulation and control studies to be performed in the same session. Using this technique, we previously reported data suggesting that broadband white noise increased rCBF significantly. In the present study, we examined the temporal course of these changes. Six normal volunteers were studied with eyes covered while CO_2 was inhaled through nasal prongs. Fourteen serial 1-min images were acquired in the following sequence: 2-min quiet, 6-min auditory stimulation, 6-min quiet. Arterial blood was automatically sampled every 18 s. The stimulation protocols replicated conditions used previously to study the physiological effects of noise [J. Meyer-Delius, Automobile Tech. J. **59**, 293 (1957)], with broadband white noise presented binaurally through earphones at 90 dBA with an on/off cycle of 30 s. Region of interest (ROI) data were obtained for at least 30 regions across 9 planes for each of the 14 time frames using a thresholding technique. Transient rCBF increases occurred in most ROI's within the first 1–3 min. Increased rCBF was also observed after the stimulation period ended, most consistently in cerebellum and paracentral regions where 10%–30% increases were seen in all subjects. These results suggest that rCBF changes with auditory stimulation have a significant temporal component that is not uniform across brain regions. This component must be defined when regional brain response to auditory stimulation is studied.

5:00

W15. Measurements of *in vitro* outer hair cell motility in the mammalian cochlea. B. Niles Evans, R. Warner, and A. Yonovitz (Speech and Hearing Institute, The University of Texas Health Science Center, Houston, TX 77030)

The sensory receptors of albino guinea pigs were dissociated and maintained in high sodium culture media. Outer hair cells were examined in response to current stimulation and cholinomimetic compounds. Two types (I and II) of movement have been detected. Small type I responses were seen when stimulated by electrodes placed in the medium along the long axis of the cell similar to those observed by Brownell (*SEM*/III, 1984). Large type I responses at higher current densities were seen by the suction electrode technique. These motile events appear to be graded in response and restricted to the supranuclear region. A video camera and phototransister were utilized with an inverted microscope. A signal averaging system allowed the recording of the displacement function of the cell. Frequency responses of the cells were derived and the input–output function modeling the movement was determined. Type II acetylcholine or carbachol-induced movements were seen in a proportion of the cells examined. These movements were slow sustained contractions which appear to be reversible when the compound is washed out of the medium. We found the outer hair cell to be exquisitely sensitive to osmotic fluctuation. Further evidence confirming efferent stimulation is presently being sought.

Session X. Shock and Vibration III: Finite Element Techniques for Acoustics

Vijay K. Varadan, Chairman

Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, Pennsylvania 16802

Chairman's Introduction—1:25

Invited Papers

1:30

X1. Discrete-element acoustic analysis of submerged structures using doubly asymptotic approximations. John A. DeRuntz, Jr. (Applied Mechanics Laboratory, Dept. 93-30, Bldg. 255, Lockheed Palo Alto Research Laboratory, 3251 Hanover Street, Palo Alto, CA 94304)

Doubly asymptotic approximations have been found to offer significant advantages for the treatment of steady-state fluid–structure interaction in vibration, acoustic-radiation, and acoustic-scattering problems for complex submerged structures. This paper describes the theoretical foundations, development, and verification of two boundary-element/finite-element processors that implement this approach. The first processor is SWEEPS, which determines the structural response of and surface pressure on a vibrating submerged body using an iterative, incremental frequency technique for computational efficiency. The second is TARGET, which embodies a discretized form of the Helmholtz integral equation to obtain fluid pressures away from the body. To test these processors, two problems involving a spherical shell in an infinite fluid have been solved. The first problem is one of modal internal forcing; while the second is concerned with forcing by incident plane waves. The computational results exhibit excellent agreement with closed form solutions.

2:00

X2. Application of finite element methods for scattering and vibration problems. Vasundara V. Varadan (Laboratory for Electromagnetic and Acoustic Research, Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

The scattering of waves by an inclusion of arbitrary shape is a problem of interest in several fields of engineering. Several of the analytical/numerical techniques that have been developed in recent years, although easy to implement and of wide applicability, are not satisfactory for truly arbitrary geometries, materials that are inhomogeneous and/or anisotropic and when nearby boundaries are involved. For such problems, we have found it convenient to develop a hybrid method using the finite element approximation in a bounded region enclosing the inclusion and the fields in the infinite exterior domain are expanded in wavefunctions that satisfy radiation conditions. Fields and appropriate derivatives are matched at physical and mathematical boundaries leading to a solution of the fields in the entire region. A slightly different approach has been developed for thin elastic shells of arbitrary shape immersed in water. For this problem the modal impedance of the shell is found using shell theory and a finite element technique. The impedance is then used in the integral representations for the exterior problem and solved by the usual *T*-matrix approach. Some of the examples that will be discussed are (1) scattering of elastic waves by voids and solid inclusions embedded in a solid, (2) scattering of elastic waves by a transversely isotropic piezoelectric inclusion, (3) scattering of acoustic waves by thin elastic shells, laminated composite shells, and stiffened shells in water, and (4) natural vibration frequencies and mode shapes of objects of arbitrary shape.

2:30

X3. Response of axisymmetric shells in an acoustic medium. A. J. Bronowicki (TRW Dynamics Department, Redondo Beach, CA 90278) and R. B. Nelson (School of Engineering and Applied Science, Department of Mechanics and Structure, University of California at Los Angeles, Los Angeles, California 90024)

The frequency domain response of arbitrary closed shells of revolution immersed in an infinite acoustic medium is considered. A reduction in dimensionality of the problem is achieved through a decomposition of motion into circumferential harmonics. The acoustic relation is thus represented as an integral equation defined along the shell meridian. This relation, derived on the basis of a Green's function technique featuring toroidal wavefunctions, is applicable to surfaces having arbitrary meridianal shape, including corners. In order to assure uniqueness of solution, interior equations are appended to the set of surface integral equations. The concept of an acoustic element is introduced with meridianal pressure variation determined by the response at a number of surface pressure nodes. The structure is represented in terms of an assemblage of conical frustum shell elements having either two or three nodal rings. The fluid–structure interaction relation is consistently developed on the basis of modal degrees of freedom. The final product is an accurate, efficient scheme to predict

structural response, and surface and exterior pressure fields. Examples considered include: spherical and toroidal substitute problems, a truncated cylinder having specified radial motion, and response of a spherical elastic shell excited by a harmonic point load.

3:00

X4. An approximation finite element surface impedance representation for energy absorbing layers. Anthony J. Kalinowski (Naval Underwater Systems Center, Smith Street, New London, CT 06320)

A procedure for determining the scattered pressure field due to a monochromatic harmonic wave that is normally incident upon an energy absorbing submerged structure is treated. The case where the structure is modeled with finite elements and the surrounding fluid is represented with either acoustic finite elements or a boundary integral approach is considered. Finite element modeling problems arise when the details of the submerged structure in the neighborhood of the fluid–structure interface are not macroscopically homogeneous and in particular when the inhomogeneities are small relative to the acoustic wavelength. An approximate procedure is presented for replacing the detailed microscopic representation of the layered surface configuration with an equivalent simple surface impedance finite element which is specially designed to work at the frequencies of interest.

Contributed Papers

3:30

X5. A finite element analysis program for reverberant spaces. Larry Sabo, John B. Ochs, and Terry Delph (Department of Mechanical Engineering and Mechanics, Lehigh University, Bethlehem, PA 18015)

A finite element analysis program has been developed to analyze the spatial distribution of pressure in irregularly shaped reverberent enclosures driven by a point source. The program uses a matrix form of the modified Helmholtz equation with a driving source term where soft wall boundary conditions are introduced with Galerkin's method to yield a complex symmetric matrix. The results compare well with the closed form solution for a point source and homogeneous absorption material on all walls over the low range of frequencies and for several absorption coefficients. However, at higher frequencies corresponding to higher eigenvalues, the low mesh density introduces error that corresponds to an increase in stiffness. Irregular shapes have also been studied. Effective use of the program for spatial distribution and frequency response analysis for forced vibration requires appropriate mesh density, adequate computer resources, and computer graphics to interpret the results. [Work sponsored by Knoll International.]

3:45

X6. Acoustic shape optimization using the finite element method. Robert J. Bernhard and Joseph R. Milner (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

Information to predict the sensitivity of acoustical response to geometric shape variations can be calculated using the acoustic finite element formulation [R. J. Bernhard, J. Sound Vib. **98**, 55–65 (1985)]. Such sensitivity information can be utilized to indicate to a designer the change of each geometric design variable necessary to achieve a better solution. The sensitivity information also indicates which design variables have minimal effect on the solution and how close the current solution is to an optimal condition. When the condition of the design can be quantified with a performance equation, numerical optimization procedures can be used to systematically find the optimal solution. Sensitivity information also permits the use of efficient gradient optimization algorithms. This paper will illustrate the incorporation of the acoustic finite element shape sensitivity information into a gradient optimization procedure. The design procedures for reactive mufflers will be illustrated. When modal techniques are appropriate for the solution of a particular problem, the shape sensitivity information may also be utilized to find modal sensitivity information.

4:00

X7. Fractal finite element mesh generation for vibration problems. J-H. Jeng, V. V. Varadan, and V. K. Varadan (Laboratory for Electromagnetic and Acoustic Research, Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802)

A new approach for solving plate vibration problems has been formulated by using the elastic Sierpinski gaskets and finite element methods. The Sierpinski gaskets are geometric fractals which are self-similar under scale transformation. Modeling a finite plate element with the basic Sierpinski gasket unit, the equations of motion of the basic unit are derived. A build up process is formed by linking three identical units together. Finally, systematic recurrence relations are generated. The advantage of this approach is that the dynamic matrix of the system has the same dimension at each build up step. The basic vibration properties, computation time, and memory requirements are discussed and critically compared with traditional finite element methods by discussing a specific example. In general, finite element mesh generation based on geometric fractals offers much promise in reducing storage requirements and computation times significantly.

4:15

X8. Application of damping elements to the modeling of underwater radiating structures. Régis Bossut, Jean-Noël Decarpigny (I.S.E.N., 41, Boulevard Vauban, 59046 Lille Cedex, France), Bernard Tocquet, and Didier Boucher (GERDSM, Le Brusc, 83140 Six-Fours-Les-Plages, France)

In the finite element method, the damping or radiating elements are often used to limit the extension of the modeling of the propagation media surrounding the radiating structure, and thus to reduce the number of fluid elements. The radiation condition is generally related to the spherical wave acoustic impedance. Recently, an improved radiating element has been developed [R. Bossut and J. N. Decarpigny, J. Acoust. Soc. Am. Suppl. 1 **74**, S23 (1983)], based on Bayliss *et al.*'s results [ICASE Rep. No. 80-1, USRA (1980)], which takes account of the dipolar contribution of the wave impedance. The work described here demonstrates that with this damping element, the F.E.M. computed pressure map, on the radiating structure and in the nearfield, is given with low error, even if the radiating surface stands deeply inside the farfield limit. Thus, a finite element analysis can provide, even with a reduced mesh, an accurate starting point for the computation of the farfield, with the help of a Helmholtz integral or an extrapolation method. Several examples are discussed.

Session Y. Speech Communication V: Speech Focus Session: Development

Patricia K. Kuhl, Chairman
Department of Speech and Hearing Sciences, University of Washington, Seattle, Washington 98195

Chairman's Introduction—1:00

Invited Papers

1:05

Y1. On attributing a "phonetic level" to infants. Patricia K. Kuhl (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

There are two distinct characterizations of young infants' linguistic ability. One account argues that infants are born with a specialized system that subserves the processing of linguistic information and that formal linguistic categories are pre-specified. This account holds that there is an "innate phonetics." The alternative account argues that young infants do not begin with formal linguistic categories. This account holds that a "phonetic level" develops at some point, guided by the infant's more general auditory abilities and cognitive strategies. An examination of the data and the arguments over the past 15 years shows that the pendulum has swung between these two characterizations. The first experiments on infants, showing categorical perception (CP) in 1-month-olds, led to a positive attribution of a "phonetic level" to infants. Later data replicating these effects in nonhuman animals removed the need to impute specialized linguistic mechanisms to infants in order to account for the CP effect. Now we have new data on infants reflecting an even more sophisticated perceptual organization of speech. These examples involve infants' abilities to categorize speech, to organize speech categories around a central "prototypical" stimulus, and to recognize cross-modal correspondences between speech information presented auditorially and visually. These data, along with our new results on animals, will be used to reconsider the initial issue about whether or not we should attribute a "phonetic level" to infants. [Work supported by NIH and NSF.]

1:35

Y2. The development of cross-language speech perception. Janet F. Werker (Department of Psychology, University of British Columbia, Vancouver, BC V6T 1Y7, Canada)

The present research was designed to determine whether infant speech perception capabilities can be explained by acoustic categorization, or whether phonetic (or at least phonetically relevant) categories must be inferred. A 16-step, synthetic place-of-articulation continuum was used. Preliminary testing showed that adult English speakers divide this continuum into two categories, /ba/–/da/, and adult Hindi speakers divide it into three categories, /ba/–/da/–/da/. Infants aged 6–8 and 11–13 months were compared to Hindi- and English-speaking adults on their ability to discriminate multiple exemplars from either side of one of three boundary locations: "Both"—the English and Hindi front/mid-boundary; "Hindi only"—the Hindi dental/retroflex boundary; and "Neither"—an arbitrary point in the "da" end of the continuum that no language uses to differentiate meaning. Results showed that young infants and Hindi adults could categorize stimuli at the "Both" and "Hindi-only" locations but not at the "Neither," but that older infants and English adults could only use the "Both" boundary. These results show the phonetic relevance of initial infant speech perception capabilities, and further demonstrate that these capabilities are reorganized during the first year of life into language-specific phonemic categories. [Work supported by NSERC.]

2:05

Y3. Representation of speech by infants. Peter W. Jusczyk (Department of Psychology, University of Oregon, Eugene, OR 97403), Jacques Mehler, Josiane Bertoncini, and Ranka Bijeljac-Babic (C. N. R. S., Laboratoire de Psychologie, Paris, France)

Infants are capable of making fine discriminations between speech contrasts. What role do such capacities play in acquiring the sound structure of a native language? One possibility is that when infants learn to recognize spoken words in their native language, they encode all of the acoustic properties they are capable of registering into their representations of words. On this view, the initial representations might be quite detailed and the effect of experience with a specific language would be to eventually eliminate those aspects not relevant to phonemic distinctions in the language. An alternative view is that infants may start with very global representations of words, sufficient to distinguish among a limited number of vocabulary items. Addition of new words would force infants to incorporate into the representations more of the information that they are capable of extracting from their perceptual analyses of speech. Experience with a specific language would serve to select those properties most relevant to making phonemic distinctions in the language. Some support for the latter hypothesis is provided by a series of experiments on newborns' and 2-month-old infants' representation of speech. [Work supported by NIH #15795 & C. N. R. S.]

2:35

Y4. Toward an acoustic typology of vocalizations in the first year of life. Raymond D. Kent (Department of Communicative Disorders, University of Wisconsin, 1975 Willow Drive, Madison, WI 53706) Harold R. Bauer (Department of Speech and Hearing Sciences, Ohio State University, Columbus, OH 43210)

Acoustic methods have been applied sufficiently to the study of infant vocalizations in the first year of life that an acoustic data base has begun to emerge in the literature. This paper considers this data base with respect to (1) its adequacy in describing the pattern of normal phonetic development, and (2) its potential for identification of infants at risk for communication disorder. With respect to (1) above, data will be summarized on several acoustic variables, including vocalic formant frequencies, utterance duration, fundamental frequency, vocal tremor frequency, noise spectra of fricative and trill segments, repetition rate of trills, and voice onset time for syllable-initial stops. With respect to impairment or palatal clefting. A preliminary acoustic typology based on an acoustic coding of infant vocalizations will be presented.

3:05–3:30

Break

Contributed Papers

3:30

Y5. Infants' discrimination of cooperating and conflicting voicing cues. Rebecca E. Eilers, Debra Moroff, D. Kimbrough Oller, and Richard Urbano (Department of Pediatrics, University of Miami, Mailman Center for Child Development, P. O. Box 016820, Miami, FL 33101)

The perceptual interaction of voicing cues that differentiate between final position /d/ and /t/ was investigated in 6–8-month-old infants. Fifty-two infants were divided into four groups; half received three contrastive pairs of two-syllable stimuli [ma + syllable] and the other half received analogous three-syllable stimuli [masa + syllable]. Infants' discrimination was tested using the visually reinforced infant speech discrimination paradigm. Half the infants in each syllable condition received cooperating cues for final voicing; the other half received conflicting cues. The cooperating stimuli were (1) [masamad]–[masamat] (voicing only), (2) [masamad]–[masama:d] (vowel duration only), and (3) [masama:d]–[masamat] (voicing + vowel duration). The competing group received (1) [masamad]–[masamat], (2) [masamat–masama:t], and (3) [masamad]–[masama:t]. Analysis of variance yielded a significant stimulus–pair effect but did not yield a significant syllable effect or a cue-group by stimulus–pair interaction. However, *post hoc* analysis indicated that vowel duration was a more potent cue than voicing and multiple cues were significantly better discriminated than single cues (pairs 1 and 2). No decrement in performance resulted from competing cues. Implications for development are discussed.

3:42

Y6. Discrimination differences for children and adults due to changes in *a priori* ratio between same and different trials. Joan E. Sussman (Department of Speech Pathology and Audiology, State University of New York, Geneseo, NY 14454)

In a previous study [Sussman and Carney, J. Acoust. Soc. Am. Suppl. 1 **75**, S45 (1984)] it was found that normal-language children demonstrated a developmental increase in sensitivity to fine acoustic cue differences of long-transition synthetic CV stimuli but not for short-transition stimuli in a discrimination task. In addition, criteria used by children differed significantly from those used by adults. An *a priori* ratio between same and different trials of 0.25:0.75 was used. The present discrimination study used the same short-transition synthetic stimuli which varied along a seven-step continuum that corresponded to a place of articulation dimension. Discrimination sets with proportions of same to different trials of: 0.50:0.50, 0.33:0.67, and 0.67:0.33 were administered to 30 children and 10 adults in individual experimental sessions. Results showed that sensitivity did change with *a priori* condition while criteria used by adults were more strict than those of children. Patterns of criteria according to *a priori* condition will be discussed. [Work supported by ASHF.]

3:54

Y7. Infant speech-discrimination testing: Effect of stimulus intensity and methodological model on estimates of performance. Robert J. Nozza (Division of Audiology, Children's Hospital of Pittsburgh, Pittsburgh, PA 15213)

Infant speech-sound discrimination data have typically been obtained at only a single stimulus intensity level, with no consideration given to the bias in outcome that can result when full performance–intensity (P–I) functions are not determined. Another potential for bias lies in the assumption made about the underlying methodological model. Data from infant head-turn discrimination studies are treated as if from a single-interval psychophysical procedure, but the infant is operating with high temporal uncertainty. The task is better described as vigilance. This investigation was done to measure performance versus intensity (50 to 70 dB SPL) in a phoneme discrimination task (/ba/ versus /da/) by infants and to determine if the choice of underlying methodological model influences results. Results indicate that performance varies with intensity and suggest caution in interpreting results obtained at a single stimulus intensity level. The P–I functions have similar form under both models, but the finding of a response decrement, typical of vigilance rather than fixed-interval tasks, supports the appropriateness of the vigilance model. Analysis of response density over time, as suggested by Watson and Nichols [J. Acoust. Soc. Am. **59**, 655–668 (1976)] for vigilance-type data, provides limited support for the use of the single-interval model with the VRISD (undefined trials) procedure. [Supported by Pennsylvania Lions Hearing Research Foundation.]

4:06

Y8. Acquisition of pitch contours by infants. Rachel E. Stark, Jennifer L. Bond, and John M. Heinz (Kennedy Institute, 707 N. Broadway, Baltimore, MD 21205)

It has been suggested that infants may acquire control of pitch contour in their vocal productions according to an invariant schedule. The present study was designed to examine pitch contours in the vocalizations of infants in the first eighteen months of life. Eight to ten infants were recorded on a limited longitudinal basis, i.e., across each of the following are ranges: 2–10, 12–23, 26–36, 40–55, and 72–88 weeks (45 infants in all). Forty to fifty vocalizations were selected randomly from each infant's output at the first and the last of these recordings. These vocalizations were analyzed acoustically by means of computer-assisted spectral analyses and pitch extraction program. Pitch contours were categorized according to the range in which they occurred, as well as their falling/rising characteristics. Developmental differences were documented with respect to pitch contour types across age levels. However, significant differences among infants within age levels were also found. The results suggest that, while there may be an invariant developmental schedule for pitch contour production, this schedule is not highly correlated with chronologic age. [Work supplied by Bureau of Community Health Services, Maternal and Child Health.]

4:18

Y9. Prosodic contours in English 2-year-olds' words. Karen E. Pollock, George D. Allen (AUS Department, Purdue University, West Lafayette, IN 47907), and Lisa Goffman (Lake County Parent Infant Center, Gurnee, IL 60031)

Polysyllabic words produced by six 2-year-old English-speaking children were analyzed for prosodic contours using wideband, narrow-band, and average amplitude spectrograms. The children were tape-recorded as they named pictures and objects during a play situation. Using the procedures previously described for the analysis of similar French children's productions [G. Allen, Acoust. Soc. Am. Suppl. 1 73, S29 (1983)], words were analyzed for fundamental frequency contours over the word, and peak intensity differences and absolute and relative vowel durations between the last two syllables. Data will be presented on each measure alone and in correlation with the other measures. Results will be discussed in relation to universal patterns of early speech production, differences between the English and French children's contours, and differences between individual subjects.

4:30

Y10. Vowel intrinsic pitch in 1-year-olds? Harold R. Bauer (Speech & Hearing Science Section, Department of Communication, The Ohio State University, 324 Derby Hall, 154 N. Oval Mall, Columbus, OH 43210)

Intrinsic pitch is distinctive of human vowels in older children and adults. Because of the ontogenetic reconfiguration of the human vocal tract, the relationship between tongue height and the fundamental frequency of infant vowels remains in question. In this paper, the instrinsic pitch of high- versus low- front and back vowels was measured from 13-month-old infants. Six infants were recorded interacting with their mothers at home. High-quality wireless microphone, time-coded, 1-h recordings were narrowly transcribed by two investigators with the aid of frequency and amplitude displays. The subsample of high and low vowels was drawn from a computerized database for acoustic analysis. Vowels were analyzed for $F0$ with narrow-band spectrographs. Measurements were made from the center of each vowel and the results subjected to statistical analysis. Results were related to individual variability in infant $F0$ and the reconfiguration of the human infant vocal tract.

4:42

Y11. Variability of timing control: Maturational or statistical? Thomas H. Crystal and Arthur S. House (Institute for Defense Analyses (CRD), Princeton, NJ 08540-3699)

Segmental durations have been studied extensively in the context of motor control and skilled action. Rather consistently it has been found that the mean durations of sounds, and associated standard deviations, produced by younger children are larger than those produced by older children and adults. This putative lengthening and increased dispersion is of interest since, if reduction of duration with age is a consequence of neuromuscular maturation, such durational measures may serve to characterize developmental progress in motor control. Kent and Forner [J. Phonet. 8, 157–168 (1980)] have pointed out, however, that "heightened variability...may really be evidence...that slow speakers are more variable...than fast speakers" independently of maturational considerations. We have examined measurements made on readings of two scripts by three fast and three slow talkers drawn from larger groups. In this corpus, as well as an earlier one [J. Acoust. Soc. Am. 72, 705–716 (1982)] the distributions of speech sounds are skewed. This suggests that the standard deviation of distributions of durational measurements does not provide an independent indication of variability that allows inferences about motor control or maturation. Sample distributions and correlations of mean values and SD's will be shown.

4:54

Y12. Acoustic consequences of fricative-vowel anticipatory coarticulation in young children. Susan Nittrouer (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The acoustic consequences of various forms of anticipatory coarticulation have been studied in adult speech, but only a few studies have investigated these effects in young children. One hypothesis is that young children coarticulate less, producing speech in a more segmental manner than adults. According to this idea, coarticulation develops gradually, possibly to improve articulatory efficiency. In the present experiment, 40 subjects (eight children at each of the ages 3, 4, 5, and 7 years and eight adults) were recorded saying the syllables /ʃi/,/si/,/ʃu/, and /su/ eight times each. The center of gravity of the fricative spectrum and second formant values, both in the fricative and at the onset of voicing, were measured. Effects of vowel context were more strongly evident in measurements made within the fricative portion of the signal for children than for adults, but children's vocalic formant transitions were reduced relative to adult transitions. These results suggest that, in fricative-vowel sequences, children tend to coarticulate (or coproduce) the two segments more strongly than adults do. This finding may indicate that, in fact, children produce speech in a less segmental manner than adults. [Work supported by NICHD Grant HD-01994.]

5:06

Y13. Developmental aspects of the production of velar stop consonants. Philip Lieberman and Joan A. Sereno (Box 1978, Linguistics Department, Brown University, Providence, RI 02912)

This experiment examines vowel context effects. In adults, the general spectral characteristics of velar stop consonants vary as a function of vowel context. A velar stop preceding a front vowel has a predominant spectral peak in the mid-frequency region whereas a velar stop preceding a back or central vowel has a spectral peak in the low-frequency region and a secondary spectral peak in the high-frequency region. In the present experiment, CV syllables ([ki], [ka]) were produced by adults and children. Short time spectra were computed for each stimulus. The results indicate that the adult stimuli, as expected, show significant vowel context effects. In the child data, however, the differences for the velar stops preceding front and back vowels are not always present. A perceptual analysis was then conducted using the aperiodic portion of the [ki] and [ka] syllables. These data show that the vowel context effects for adults are perceptually salient but that these cues are often not perceptible in the child stimuli. The results are discussed in terms of their implications for language learning.

5:18

Y14. Plasticity in adult and child speech production. James Emil Flege (Department of Biocommunication, University of Alabama at Birmingham, University Station, Birmingham, AL 35298)

This study examined the ability of nine adult and nine children (aged 6–9 years) to speak with thin (0.5 mm) experimental prostheses in the mouth. One prosthesis conformed closely to the upper teeth and hard palate; the other substantially altered oral cavity geometry by reducing by one-half the volume under the palatal vault. Degree of adaptation was estimated by having listeners rate the adequacy of words (*pea, tea, sea, she*) produced in two lists read before insertion of the prostheses (baseline), immediately after insertion, 4 and 12 min after insertion, and twice after removal. The children were less able than adults to adapt to modest vocal tract changes, but more able to adapt to changes requiring reparameterization that went beyond the normal bounds of articulatory variation. Ratings of the children's but not the adults' production of *sea* decreased significantly compared to baseline immediately after insertion of the conforming prosthesis. However, the children showed somewhat greater success in adapting to the cavity-altering prosthesis, the presence of which caused a sharp drop in listeners' ratings. Their *sea*, but not the adults', received significantly higher ratings when produced 12 min than immediately after insertion. These findings suggest that the developing coordinative structures of children are less *flexible* but more *plastic* than those of adults. [Work supported by NIH grant NS20572.]

WEDNESDAY AFTERNOON, 14 MAY 1986 — WHITE ROOM, 1:30 TO 5:05 P.M.

Session Z: Underwater Acoustics IV: Signal Processing in Underwater Acoustics

Terrence L. Henderson, Chairman
Applied Research Laboratories, University of Texas, Austin, Texas 78713

Chairman's Introduction—1:30

Contributed Papers

1:35

Z1. Transmission intensity in shallow oceans, and implications for passive tracking. J. S. Robertson (U. S. Military Academy, West Point, NY 10996-1786), M. J. Jacobson, and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

The behavior of incoherent total-field intensity is considered for a moving cw source in shallow water. The ocean channel is assumed isospeed, with a planar perfectly reflecting surface and a planar lossy bottom. The source is located at constant depth on a linear path, and the receiver is fixed on the bottom. An expression for incoherent total-field intensity is derived in terms of source-motion and environmental parameters. For a bottom with uniform loss per ray reflection, an approximation for intensity using special functions is developed and analyzed. The behavior of this novel approximation with respect to variations in source parameters such as range, depth, and speed, as well as bottom loss variations, suggests its usefulness for prediction of source motion. Two properties of the approximation, peak curvature and peak width, are selected as source motion descriptors. Both are shown to be relatively sensitive to changes in source parameters and insensitive to bottom loss variations. We apply our descriptors to an example, and discuss their use in supplementing existing passive tracking methods. [Work supported by ONR.]

1:50

Z2. Spectral broadening by ocean microstructure. Lou Goodman and Adam Jilling (Naval Underwater Systems Center, New London, CT 06320)

In the past decade there has been a great deal of interest in the use of a sonar to infer physical properties of the ocean. Acoustic Tomography and FLIP high-frequency active sonar system are examples of successful efforts. Recently there has been interest in whether high-frequency sonar can be used to measure the smallest scale of ocean fluctuations, microstructure, typically of order meters and smaller in the open ocean. A critical problem which arises in attempting such measurements is that the scattering volume is typically of order of the scale of the variability resulting in Doppler spread as well as Doppler shift. In this work the nature of this spread is examined using a realistic form of microstructure field—one which takes into account both space and time variability. A Brownian motion type of model for the scatterers, which is often employed, will be shown in general to be inappropriate. It will be shown that the temporal correlation of the returned signal is not ergodic. This result is related to the nature of the fourth-order moment, which unlike the second-order moment contains information on the spatial correlation of the scatterer motion.

2:05

Z3. Measurements of the source levels of shallow explosive charges. N. Ross Chapman (Defence Research Establishment Pacific, FMO Victoria, BC V0S 1B0, Canada)

Experiments have been carried out to measure the source levels of standard 0.82 kg SUS (Signal Underwater Sound) charges at nominal depths of 18, 91, and 180 m. The source levels are reported in this paper for 1/3 octave bands from 5 to 500 Hz. Approximately 30 charges for each depth were included in the analysis, and the standard deviation was less than 1 dB for each frequency band. The source levels estimated from these data were compared to the previous results of Gaspin and Shuler which were based on hand drawn pressure waveforms. Their values are generally within 2 or 3 dB of the measured source levels but there are some greater differences. Reasons for the disagreement between the two sets of values are suggested. In addition to the measured source levels, source levels for 0.82 kg SUS at 244 and 610 m, and for 22.7 kg (50 lb) charges at 100 and 200 m have been estimated using a scaling technique [R. Hughes, in *Underwater Acoustics and Signal Processing* (Reidel, New York, 1981), pp. 87–91]. The spectra of standard SUS measured at 91 and 180 m were scaled, and the source levels obtained in this way are accurate for frequencies up to three or four times as large as the bubble frequency. These results are also presented in 1/3 octave bands, and are compared to previously published values.

2:20

Z4. Eigenvalue/eigenvector characterization of slope reflected signals. E. Livingston and A. Tolstoy (Naval Research Laboratory, Code 5120, Washington, DC 20375-5000)

Eigenvector/eigenvalue analysis of experimental acoustic propagation data obtained in the region of the East Australian Continental Slope indicates this method is sensitive to changes in multipath structure and can distinguish multiple signal arrivals. A 16-element horizontal array at 300 m was towed in deep water, while a 152-Hz source at 18-m depth was moved from shallow to deep water. Source–receiver separation was about 60 nmi with source bearing near array endfire. Shipboard processing consisted of 1024-Hz time sampling, 4K FFT processing, and production of

WEDNESDAY PM

cross-spectral matrices for selected frequency bins around 152 Hz. The eigenvalues of a spatially whitened cross-spectral matrix may be used to count the number of statistically independent arrivals. These may be from multiple sources or from multipath effects of a single source. The eigenvalues for each sample cross-spectral matrix (five minute averages) were calculated and displayed versus time in order to correlate the number of large eigenvalues with the number of signal arrivals. In addition, conventional, minimum variance (ML), and eigenvector beamforming produced wavenumber-time displays of signal arrivals. For comparison, a GRASS-based ray trace model, using actual bottom reflection coefficients and deep water sound-speed profile, was used to predict expected arrivals. Comparisons of the visible shift in eigenvalues with arrivals in both the beamformed and modeled data show the eigenvalue detection method to be sensitive to changes in the multipath structure and capable of distinguishing multiple signal arrivals that were not detectable with conventional beamforming methods.

2:35

Z5. Using a multi-dimensional matched filter algorithm to separate multipaths. S. A. Reynolds, B. A. Bell, and T. E. Ewart (Applied Physics Laboratory, University of Washington, Seattle, WA 98105)

Acoustic pulse arrivals can often be modeled as a sum of time-shifted, amplitude-scaled replicas. When these arrivals occur at separations of a few wavelengths or less, a multi-dimensional method is required to separate the individual paths. An algorithm that separates the paths by globally optimizating a multi-dimensional matched filter (Bell and Ewart, submitted to IEEE Trans. Acoust., Speech, and Signal Process.) has been used on two data sets. One set, taken during the Mid-ocean Acoustic Transmission experiment (MATE), is heavily contaminated by multipath. The second, taken during the AIWEX Acoustic Transmission Experiment (AATE), is a very large data set that contains segments contaminated by closely spaced paths. The advantages of this algorithm over previous ones are speed and the pulses are solved independently. Because the algorithm is fast, several passes through the data are possible. By varying the dimension of the filter, a better picture of the temporal structure of the multipath arrivals is possible. The effect of errors have been examined.

2:50

Z6. Source parameter estimation in correlated spatial noise. A. B. Baggeroer (Massachusetts Institute of Technology, Cambridge, MA 02139), W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375), and H. Schmidt (SACLANT ASW Research Center, I-19026, La Spezia, Italy)

Several authors have proposed estimating the location parameters of a source using coherent predictions of the acoustic field. The signal processing is the spatial equivalent to matched filtering which is optimum only for white noise, i.e., uncorrelated from sensor to sensor. Alternatively, others have introduced the maximum likelihood method which can be shown to be a simple mapping to the matched filter if the noise is white. For closely spaced sensors white noise is not a good model; moreover, the signal and noise propagate in the same medium and must have similar characteristics. (For example, in the ocean waveguide model both the signal and noise are functions of the shape and composition of the boundaries as well as the sound-speed profile.) This paper introduces the Kuperman–Ingenito correlated noise model [J. Acoust. Soc. Am. 67, 1988 (1980)] to obtain predictions of source location parameter estimation performance. These are compared with the appropriate Cramer–Rao bounds.

3:05

Z7. Inferring the sound-speed profile from shots recorded on vertical arrays. L. Neil Frazer (Hawaii Institute of Geophysics, 2525 Correa Road, Honolulu, HI 96822)

Let $\phi(x,z,\omega)$ be the pressure data recorded on a vertical array from a shot at range x. Suppose there are N such shots, all at the same depth but with a range increment Δx. The resulting pressure data are used to form the data matrices A and B in each of which the row index is depth and the column index is range. The first (last) column of A is the data from the second (last) range whereas the first (last) column of B is the data from the first (next to last) range. Therefore, $A = \exp(i\Delta x H)B$, where H is the matrix of the operator $[\omega^2/c^2(z) + \rho(z)\partial_z\rho^{-1}(z)\partial_z]^{1/2}$ in the PE equation $(i\partial_x + H)\phi = 0$ governing pressure propagation in a stratified ocean. To get the sound-speed profile first form $H = (1/i\Delta x)\ln(AB^-)$, where B^- is a generalized inverse of B; then multiply H by itself to get H^2; then compute the "standard symbol matrix" $H^2_s(p,z) = \int dz' \exp(ipz') \times H^2(z',z'-z)$; then fit a quadratic in p to each column of the symbol matrix in order to recover the coefficients of p^0 and p in the expression $H^2_s(p,z) = \omega^2/c^2(z) - p^2 - ip\rho'(z)/\rho(z)$. [Work supported by ONR.]

3:20

Z8. Approximation of the dynamics of a towed sonar antenna using linear FIR filters. Wulf Brandenburg (FGAN-Forschungsinstitut für Hochfrequenzphysik, Abteilung SuK, Neuenahrer-Strasse 20, D-5307 Wachtberg-Werthoven, West Germany)

In low-frequency sonar systems, long towed arrays are used to estimate target parameters like bearing etc. The standard method for bearing estimation is based on conventional beamforming. This procedure requires the knowledge of the sensor locations. Therefore, a point mechanical model for the dynamics of a towed array is developed, which is described in the Proceedings of the IEEE-ICASSP, 1984. These results show that the dynamics of a towed array can be approximated by linear systems with transfer functions given by polynomials (FIR filters). The advantage of this method is that the shape of the towed array can be easily estimated in real time by convolution while the tow ship is performing maneuvers. One needs the system functions which are only calculated once and then stored in a computer. For each point of the towed array there exists a different system function. The input of the system "towed array" is the course $\Phi_0(t)$ of the towing ship at time t and the outputs are the directions $\Phi_i(t)$ of the tangential vectors at different points $i = 1,\ldots,N$ of the towed array. This method is very useful, for example, for adaptive beamforming or for evaluation of the sensitivity of the beam pattern to the two ship's maneuvers. Examples show that the approximation is valid for a wide range of parameters. This is also verified by measurements.

3:35

Z9. Implementation of the Wigner distribution function for representing the dynamics of acoustic signal spectra. N. Yen (Acoustics Division, Physical Acoustics Branch, Naval Research Laboratory, Washington, DC 20375-5000)

The Wigner distribution function, originally defined in quantum mechanics for the probability function expressed in terms of the simultaneous value of coordinates and momenta, is used to represent the time-varying signal cojointly in time and frequency domains. Although its formulation resembles the Fourier transform of the time-dependent correlation function, direct application of the transformation technique to its discrete version for a finite duration signal generally encounters problems of aliasing and nonpositive value during computation. To alleviate those shortcomings, a numerical process based on the FFT algorithm has been developed by reformating the given signal and the computed result. This modified Wigner distribution function provides the dynamic interpretation for the signal's frequency component change with time. Examples of its implementation with several types of acoustic signals are illustrated wth such time–frequency representation.

3:50

Z10. *In situ* optimal reshading of arrays with failed elements. M. S. Sherrill and R. L. Streit (Code 3232, Naval Underwater Systems Center, New London, CT 06320)

An algorithm is presented which computes optimal weights for arbitrary linear arrays. The application of this algorithm to *in situ* optimal reshading of arrays with failed elements is discussed. It is shown that optimal reshading can often regain the original sidelobe level by slightly increasing the mainlobe beamwidth. Three examples are presented to illustrate the algorithm's effectiveness. Hardware and software issues are discussed. Execution time for a 25-element array is typically between 1 and 2 min on a HP9836C microcomputer.

4:05

Z11. Effects of ocean environment on the operation of a synthetic aperture sonar. Jean-Hwan Tarng and C. C. Yang (Department of Electrical Engineering, The Pennsylvania State University, University Park, PA 16802)

The application of a synthetic aperture radar has made the resolution increase tremendously in microwave imaging on the ground. The same principle can be applied to acoustic imaging for targets in an ocean. However, the scenario of acoustic wave propagation in the ocean environment is much more complicated than that of microwave propagation in the atmosphere. In this paper, effects of ocean environment on the operation of a synthetic aperture sonar are discussed. The sound-speed profile is the first factor whose effects need to be investigated. Also, random fluctuations of the sound speed due to fine structures and internal waves in the ocean produce other effects. Among the parameters of a synthetic aperture sonar, the sizes of the resolution cell in the azimuthal and range directions, azimuth ambiguity, and range ambiguity are of great concern. Cases with various sonar carrier velocities are considered. [Work supported by ARL, The Pennsylvania State University.]

4:20

Z12. Peak sidelobe estimator for randomized arrays. James C. Lockwood and David Law (Surveillance Department, Naval Ocean Systems Center, San Diego, CA 92152-5000)

A randomized line array is formed by randomly perturbing an equally spaced, deterministic array. For small perturbations, it has been shown that the variance of the sidelobe amplitude is less, over a significant portion of the beam pattern, than for larger perturbations. The present paper quantifies the effect of this variance dependency on the magnitude of the peak sidelobe. The analysis follows that presented by Steinberg [*Principles of Aperture and Array System Design* (Wiley, New York, 1976), pp. 147–166], in which for a given confidence level the upper bound on the peak sidelobe is derived. Steinberg's work is based on a totally random array, and makes use of the fact that all the sidelobes are statistically the same to obtain a closed-form expression for the peak sidelobe level as a function of confidence level. In the present work the mean and variance associated with each independent sidelobe position are computed and combined to give results that depend on the original array configuration and on the size of the perturbation. It is shown that a randomized array with a moderately small perturbation may have a significantly lower peak sidelobe than the corresponding totally random array.

4:35

Z13. Method for study of the three-dimensional beam patterns and directivity indices of volumetric arrays using a microcomputer. David C. Greene (Lexitech, 2289 Dosinia Court, Reston, VA 22091)

PASCAL language computer programs are described which allow the calculation of the three-dimensional sensitivity patterns and directivity indices (DI's) of volumetric arrays of omnidirectional receivers steered to any direction in space. The method reads a disk file containing the array design (element positions and real and imaginary shading coefficients), responds to user inputs of frequency, beam-steer direction, and angular resolution (in colatitude and azimuth angles), and produces a disk file containing the response relative to that at the steered direction for each resolved element of solid angle. The output disk file is accessed by a separate program to plot selected beam patterns. The directivity index for the array at the specified frequency and steer direction is estimated by numerical integration and presented on the console. The number of receiving elements in the array and the resolution in solid angle are limited only by available memory (5-deg × 5-deg resolution over all space on the author's implementation).

4:50

Z14. Low-frequency transmission losses and effective flexural rigidity of cracked ice cover. 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, BC V0S 1B0, Canada)

Upward refraction in the artic causes substantial interactions of underwater acoustic waves with the ice cover. Although several research papers focused on the effects of ice cover roughness on transmission losses, apparently no studies have been reported on the effects of cracks in the ice cover on reducing the flexural rigidity of the ice plate in relation to leaky Lamb waves and energy exchange between the water and the ice. Coupling between the acoustic waves in the water and the elastic waves in the ice plate depends on the relative densities and phase velocities of the waves. Depending on the number, size, and spatial distribution of the cracks, the flexural wave velocity of a low-frequency component may be higher or lower than the water compressional velocity. The steep sloped region of the flexural wave dispersion curve makes the low-frequency transmission characteristics very sensitive to the presence of cracks which reduce the flexural rigidity. Laboratory ultrasonic models are used to provide physical insights into a potentially important low-frequency transmission loss mechanism caused by the effective reduced flexural rigidity of the cracked ice cover. Thin plates with machined multiple shallow cracks are utilized. The crack spacing is about half a wavelength and the crack depth is less than 1/6 the plate thickness. [Work supported by DREP.]

WEDNESDAY PM

Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

Technical Advisory Group for ISO/TC 43/SC1 Noise

K. M. Eldred, Chairman S12
P.O. Box 1037, Concord, Massachusetts 01742

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC1
Biodynamics & Bioengineering Division, AFAMRL/BB U.S. Air Force, Wright-Patterson AFB, Dayton, Ohio 45433

Working group chairpersons will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC1 activities will be discussed.

THURSDAY MORNING, 15 MAY 1986 RITZ ROOM, 9:00 A.M. TO 12:15 P.M.

Session AA. Engineering Acoustics II: Ultrasonic Sensing in Air—Lectures

Robert Hickling, Chairman
Engineering Mechanics Department, General Motors Research Labs., Warren, Michigan 48090-9055

Mauro Pierucci, Vice Chairman
Department of Aerospace and Engineering Mechanics, San Diego State University, San Diego, California 92182

Invited Papers

9:00

AA1. Very-high-resolution object imaging by CTFM airborne echo location. Leslie Kay, Christopher Anderson, Michael J. Cusdin, and Julian Sinton (Department of Electrical & Electronic Engineering, University of Canterbury, Christchurch, New Zealand)

An ultrasonic unitary transducer array incorporating 60 independent radiating elements has produced a focused beam of airborne ultrasonic waves varying in wavelength from 1.5 to 3 mm as the frequency varied cyclically from 200 to 100 kHz. The focal region was digitally controlled in four radial steps within the range 0.3 to 0.6 m, and in 60-deg steps. Measurements showed the resolution in radial distance to agree closely with the theoretical limit of 1.6 times the mean wavelength of 2.25 mm. Azimuth resolution agreed well with the theoretical value when using an aperture of 150 mm and a focal distance of 0.3 to 0.6 m. This was measured to be 9 mm at a radial distance of 0.45 m. Resolution is here defined as the displacement necessary to reduce the peak intensity by 20 dB. The cross section of a 120-mm cylinder covered with corrugated paper to produce spaced specular reflections was imaged showing the resolution limits. The CTFM system was highly noise resistant compared with a pulse system having the same radial resolution. This is because to achieve an octave bandwidth, solid dielectric capacitance transducers are used. The maximum pulse power of approximately two cycles duration is therefore limited to the maximum continuous wave power as used in the CTFM system.

9:25

AA2. The use of acoustic versus optic range sensors in manufacturing systems. F. B. Prinz (Department of Mechanical Engineering, Robotics Institute, Carnegie-Mellon University, Pittsburgh, PA 15213)

Range information is very important for automatic control of manufacturing systems. Part location, edge detection, and motion guidance are just a few examples of problems in which range information is required. Through simple time-of-flight measurements acoustic systems can conveniently and reliably provide the distance between the sensor and the workpiece. Scanning with the acoustic transducer furthermore allows one to reconstruct the nearfield 3-D environment. Difficulties arise primarily from the need that the workpiece surface should be reasonably parallel to the surface of the acoustic transducer. Iterative search strategies allow one to overcome this problem. Alternatively, optical range-finding systems typically use triangulation strategies to measure the sensor/workpiece distance. Their depth resolution is comparable to that of acoustic transducers. The lateral resolution is better in optical systems. Furthermore, the orientation dependence of the workpiece/sensor surface is less critical than it is for acoustic transducers. Acoustic systems are less expensive than their optical counterparts. They also can survive in the sometimes harsh manufacturing environment. The advantages and disadvantages of optical and acoustical range finding sensors are discussed in the context of applied manufacturing problems.

9:50

AA3. Precision in ultrasonic gauging. Robert Hickling (Engineering Mechanics Department, General Motors Research Laboratories, Warren, MI 48090-9055) and Samuel P. Marin (Mathematics Department, General Motors Research Laboratories, Warren, MI 48090-9055)

In ultrasonic gauging it is important to realize that only the perpendicular distance to a surface can be measured unambiguously using a single emitting–receiving transducer. Factors that affect the accuracy of the measurement are, signal-to-noise ratio, spot size, depth of field, alignment of the transducer axis perpendicular to the reflecting surface, dimensional accuracy in the construction of the transducer, and compensation for change in the speed of sound in air due to temperature changes and other effects. Experimental results are presented which show that distances up to 200 mm can be measured with an accuracy of ± 0.1 mm, provided there is adequate compensation for change in temperature of the air. Spot size, or lateral resolution, is dependent on the frequency of the ultrasound which has an upper limit of about 1 MHz because of attenuation due to absorption. The smallest spot size that can be attained, therefore, is about 1 mm, between the 6-dB points. However, a deviation of 1 deg from the perpendicular can produce a lateral error of about ± 2 mm at a distance of 100 mm. Factors affecting accuracy in ultrasonic gauging are reviewed in the paper.

10:15

AA4. Ultrasonics in manufacturing. J. A. G. Knight, S. C. Pomeroy (Department of Production Engineering and Production Management, University of Nottingham, University Park, Nottingham, NG7 2RD, UK), and R. L. Beurle (Department of Electrical and Electronic Engineering, University of Nottingham, NG7 2RD, UK)

Research was initiated at the University of Nottingham, UK in 1983 to develop ultrasonic sensor systems for flexible manufacturing. This research work, undertaken jointly by the departments of Production Engineering and Electrical Engineering, has developed into a research center for investigating the uses of ultrasonics in manufacturing. Industrial participation has been present from the inception of the research through a cooperating partner, Transfer Technology Ltd. The industrial contribution to the research is shortly to be enhanced through the formation of a "club" of industrial companies, all of which will commit resources, and in return will receive access to the research findings so as to be able to implement ultrasonics into manufacturing more effectively. The research to date has developed both single transducers and arrays to operate at frequencies up to 200 kHz in air. A computerized automatic test facility has been built for transducers. A range measuring system has been constructed utilizing two proprietary ranging systems interfaced to a microcomputer. The system operates between 130 mm and 10 m and has a range resolution of 0.05 mm at distances up to 500 mm. An ultrasonic imaging evaluation system has been built consisting of a dedicated multichannel acquisition system attached to a microcomputer. Image formation and analysis are performed in software. Further research involves the integration of ultrasonic information with simple binary vision information to obtain high resolution 3-D information.

10:40

AA5. Ultrasonic background noise in industrial environment. Henry E. Bass and Lee N. Bolen (Physical Acoustics Research Laboratory, Department of Physics and Astronomy, The University of Mississippi, University, MS 38677)

Special high-frequency microphone systems were developed, and ultrasonic noise associated with various manufacturing operations was measured in the frequency range 0–1 MHz in order to evaluate typical noise environments for ultrasonically controlled robots. Industrial operations studied include impact, bending, grinding and drilling, laser etching, and high-velocity fluid or air sprays. The first three operations provide little ultrasonic energy above 100 kHz. For these processes, ultrasonic sensors operating above 100 kHz should have an advantage since signal and noise will both fall off with increasing frequency. Ultrasonic emission from laser etching was found to be quite broadband, making such processes less suitable for ultrasonic control. High-velocity fluid or air sprays are most prevalent sources of ultrasonic noise. The noise from these sources rolls off slowly with frequency, and no frequency bands without noise exist for ultrasonic sensing below several hundred kHz. For applications requiring ultrasonic sensing, aerodynamic noise should be reduced.

11:05

AA6. Effects of airborne upper sonic and ultrasonic acoustic radiation and proposed threshold limit values for human exposure. Paul L. Michael (Environmental Acoustics Laboratory, The Pennsylvania State University, 3 Moore Building, University Park, PA 16802) and Roger L. Kerlin (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

In 1974, a literature search and a field study by these authors and co-workers at the Environmental Acoustics Laboratory were completed and sought to evaluate industrial exposure to airborne acoustic radiation above 10 kHz. The reported effects of exposure to such radiation and the three existing sets of human exposure limits for airborne upper sonic and ultrasonic acoustic radiation proposed by investigators from three different countries (the USA, England, and Russia) were documented and discussed as part of a final report prepared for the sponsor [P. L. Michael, R. L. Kerlin, G. R. Bienvenue, and J. H. Prout, *An Evaluation of Industrial Acoustic Radiation Above 10 KHz*, Environmental Acoustics Laboratory, The Pennsylvania State University, University Park, PA, Final Report, Contract No. HSM-99-72-125, Physical Agents Branch, NIOSH, Cincinnati, OH, p. 210 (February 1974)]. These results will be presented and discussed, and, in particular, their role as part of the documentation for the threshold limit values for airborne upper sonic and ultrasonic acoustic radiation in the work environment proposed by the American Conference of Governmental Industrial Hygienists. [Work supported by NIOSH.]

Contributed Papers

11:30

AA7. Noncontact ultrasonic gauging for industrial applications. John K. Billings (Ultrasonic Arrays, Incorporated, Woodinville, WA 98290)

Ultrasonic arrays (UAI) have made advances in air-coupled, pulse-echo acoustic technology which permit noncontact gauging for many industrial applications. Noncontact ultrasonic dimensional gauging applications include measurement of thickness, distance, position, perpendicularity, shape, and orientation. Accuracy is the greater of plus or minus 0.1% of range or 0.001 in. over a dynamic range of 0.2 to 23.999 in. UAI's accuracy improvement is due to the development of an advanced-technology capacitive transducer that transmits and receives a high-frequency unambiguous sound pulse. In addition, environmental factors that affect the speed of sound in air have been largely eliminated through extensive signal processing and an external reference that recalibrates target data during each sample. The microprocessor-based controller and external reference are also used for automatic gain control thereby preserving the integrity of the waveform over time and in varying environments. The controller provides communications with other devices via RS 232 and RS 422 serial data ports as well as multiplexing multiple transducer channels. On-board software permits the selection of sample rates for rapidly moving targets, high and low limits, relay configuration, echo selection, target filtering, and diagnostics. Both 4- to 20-mA current and 0- to 10-V dc current are available for control functions. Other requirements for gauging include the use of parabolic mirrors for focusing, tilting mirrors for sound directional changes, and shaded apertures for surface detail in cluttered backgrounds.

11:45

AA8. High-frequency acoustic systems for robotic applications. R. W. Smith, Rex Walters, John Carlson, and Robert Harris (Staveley NDT Technologies, 421 North Quay, Kennewick, WA 99336)

High-frequency ultrasonic systems, operating in the 200-kHz to 1-MHz frequency range and using air as the acoustic coupling medium are an effective complement to optical systems for robotic and industrial measurement applications. An air-coupled ultrasonic system was developed which produces well damped and focussed ultrasonic beams with good spatial resolution. Tests show that distance measurements with a resolution of 0.001 in. can be made through several inches of air at a rate of several hundred Hz. Focused transducers can reduce the effective area of measurement to below 0.05 in. at the focal point. The system was employed in several applications, including parts profiling, thickness testing, and the detection of missing parts. The apparent advantages that acoustic devices have over optical systems include lower cost by a factor of 5 or 10, and a higher speed of operation by perhaps 2 orders of magnitude.

12:00

AA9. Distance measurement using airborne sound. Paul A. Meyer (Krautkramer Branson, P. O. Box 350, Lewiston, PA 17044)

Inspection of manufactured material and components often involves characterization of size or shape. If large quantities of components must be inspected, as is usually the case on a production line, a rapid automated inspection process is desirable. This presentation reviews the use of airborne ultrasound as the basis for such a system. Time-of-flight measurements of acoustic pulses through air can be used for distance measurement with range resolution of 0.01 mm (0.0004 in.). Coupling this instrumentation to a data acquisition system permits high-speed, automated dimensional checks. But a more important feature of an airborne sound system is that contacting the part is not required enabling rapid characterization of components difficult to measure by other means. These include runout on abrasive surfaces such as grinding wheels or discontinuous surfaces, such as tire treads. Finally, since ultrasonic distance measurements can be updated at several hundred hertz noncontact time motion studies are possible. No loading is applied to the structure under investigation so that characterization of "delicate" vibrating components is possible.

Session BB: Physical Acoustics VI: Propagation: Solids and Fluids

Walter G. Mayer, Chairman
Department of Physics, Georgetown University, Washington, DC 20057

Contributed Papers

8:30

BB1. Ultrasonic study of an emulsion of toluene in water. M. A. Barrett Gultepe, M. E. Gultepe, J. L. McCarthy, and E. Yeager (Ultrasonic Research Laboratory, Department of Chemistry, Case Western Reserve University, Cleveland OH 44106)

Previously reported ultrasonic attenuation results in the frequency range 4 to 50 MHz and theoretical predictions [J. R. Allegra and S. A. Hawley, J. Acoust. Soc. Am. **51**, 1545 (1971)] for an emulsion of toluene in water have indicated that the observed excess attenuation in the measured frequency range was largely due to heat conduction losses. In the present paper, ultrasonic absorption measurements on a 20% emulsion of toluene in water in the frequency range 1 to 135 MHz at 25 °C are described. The excess attenuation due to the emulsion globules as a function of frequency was compared with explicit expressions for heat conduction, viscous drag losses, and scattering of sound by emulsion globules. Unlike the previous work, the agreement between theoretically predicted excess attenuation and that observed can only be obtained if a wide distribution of globule sizes was assumed. The assumption of a globule size distribution spanning from 600 to 6000 Å in diameter produced a theoretical curve in good agreement with the observed excess absorption over the entire frequency range. [Work supported by ONR.]

8:45

BB2. Ultrasonic study of concentrated coal–water slurries. M. A. Barrett Gultepe, M. E. Gultepe, and E. Yeager (Ultrasonic Research Laboratory, Department of Chemistry, Case Western Reserve University, University Circle, Cleveland, OH 44106)

The use of ultrasonic absorption measurements as a diagnostic tool for monitoring the particle size of coal–water slurries has been suggested by M. C. Davis [J. Acoust. Soc. Am. **65**, 387 (1979)]. The present paper reports some experimental data obtained from ultrasonic absorption and velocity measurements on electrostatically and sterically stabilized coal–water slurries of average particle diameters of 5 and 50 μ. Ultrasonic absorption measurements were made in the frequency range of 1 to 25 MHz for 1 °C to 80 °C. The results are compared with various explicit expressions for heat conduction, viscous drag losses, and scattering of sound by particles. The experimental absorption was consistently higher than that predicted, even when allowing for the averaging effect of a wide particle size distribution. The experimental results also indicate that velocity measurements in a coal–water slurry have the potential of becoming another possible diagnostic tool along with attenuation measurements. [Work partially supported by ONR.]

9:00

BB3. Vapor effects on bubble oscillations A. Prosperetti (Department of Mechanical Engineering, The Johns Hopkins University, Baltimore, MD 21218)

A simplified mathematical model for the description of the thermofluid dynamical behavior of a bubble containing gas and vapor is described. The pressure in the bubble is taken to be spatially uniform. Diffusion effects and evaporation–condensation processes are properly accounted for in the framework of this approximation. The model is applied to the study of several aspects of forced oscillations which are examined for different temperatures of the hot liquid. The implication of the results for the propagation of pressure waves in bubbly liquids is briefly considered.

9:15

BB4. Sound wave propagation through a polydispersion of rigid particles. Timothy S. Margulies (U. S. Nuclear Regulatory Commission, Washington, DC 20555) and W. H. Schwarz (The Johns Hopkins University, Baltimore, MD 21218)

The theory for infinitesimal, planar-sound wave propagation in a dilute, polydispersion of rigid spherical particles suspended in a viscoelastic fluid has been developed. Particle–fluid interaction terms for both momentum and heat transfer are taken into account in the linearized average continuum equations. General results are obtained in the form of a biquadratic characteristic equation for the complex propagation constant $\chi = (\alpha + i\omega/c)$, where α is the attenuation coefficient, c is the phase speed of the progressive wave, and ω is the angular frequency. These results represent a generalization of those previously derived by S. Temkin and R. A. Dobbins [J. Acoust. Soc. Am. **40**, 317 (1966)] to viscoelastic fluids. Also, the drag force on each particle includes modified Stokes drag, added mass, and Basset (time history) terms appropriate for a simple fluid with memory. Computer simulations have been made indicating the dependence of the observables, α and c, upon physical properties, such as the acoustic viscosity, particle size/distribution, and solid volume fraction, as well as, upon the drag forces modeled. Similarities and discrepancies among results of different theories, for example those by Urick and Allegra-Hawley, are discussed using available experimental measurements.

9:30

BB5. Re-examination of boric acid relaxation with high-precision spherical resonator. M. A. Barrett Gultepe, M. E. Gultepe, E. Yeager (Ultrasonic Research Laboratory, Department of Chemistry, Case Western Reserve University, University Circle, Cleveland OH 44106), and R. H. Mellen (PSI Marine Sciences, New London, CT 06320)

The dependence of low-frequency sound absorption on *p*H as a function of $B(OH)_3$ concentration in 0.725M NaCl was studied with a computer-controlled high-precision spherical resonator. For these measurements, a 100-liter titanium spherical resonator was mode mapped in the frequency range of 2 to 100 kHz when containing distilled water, then 0.1M, and finally 0.725M NaCl solutions. Frequency shifts and decay rates of the radial modes were studied with the NaCl concentrations. Using a solution of boric acid equal in concentration to sea water (4.2×10^{-4} M) at *p*H 8.2, in 0.725M NaCl, reliable absorption measurements have been made at frequencies as low as 6 kHz. [Work partially supported by ONR.]

9:45

BB6. Factors affecting the ultrasonic measurement of porosity in cast aluminum. David V. Rypien and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

An ultrasonic method was developed to measure the average pore radius and volume fraction of porosity in cast aluminum [L. Adler, J.

Acoust. Soc. Am. Suppl. 1 76, S83 (1984)]. This study was limited to samples having smooth parallel surfaces, containing average pore radii ranging from 50 to 200 μm, and volume fractions less than 6%. In order to increase the practical applications of this method, samples were analyzed with: (1) the as-cast (rough) surface intact, (2) with increased volume fractions, and (3) increasing the length of the cast samples. By using the original method, it was found that the calculated volume fraction for the above cases were significantly underestimated. Analyses of these factors and methods for determining possible corrections to improve this method are suggested. The results obtained from this investigation are also compared to the previous measurements. [This work was supported by the Center of NDE operated by Ames Laboratory for the Air Force Wright Aeronautical Materials Laboratory.]

10:00

BB7. Wideband acoustic response of fluid-saturated porous rocks. Ehud J. Schmidt and Amos M. Nur (Rock Physics Project, Geophysics Department, Stanford University, Stanford, CA 94305)

The complex compressional and shear frequency-dependent elastic constants of nonporous solids and porous rocks are measured in the frequency range 5 kHz to 1 MHz. The measurements are performed in a continuous wave acoustic transmission bridge using cylindrical samples. Use of waveguided samples prevents problems of diffraction and wavefront spreading, which are difficult to correct over broad frequency ranges. The theory of waveguided elastic wave propagation in media with complex elastic constants (generalization of the Pochammer–Cree solution) is presented. This theory is utilized in the interpretation of the experimental results from nonporous solids with real and complex frequency-independent elastic constants, solids with complex frequency-dependent elastic constants, and rocks with effective complex frequency-dependent elastic constants. For solids with frequency-independent complex elastic constants (constant Q) the results reveal the appearance of specific dissipation peaks associated with waveguide geometry. Results for fluid-saturated porous rocks are discussed in the context of the cumulative effects of fluid viscosity, solid–fluid interfacial energy, and grain scattering.

10:15

BB8. Model for a linear viscoelastic medium that has consistent creep and stress-relaxation properties. Anthony J. Rudgers (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337)

The stress and strain tensors for a homogeneous isotropic viscoelastic material are related by a general linear transformation, which expresses a complete proportionality of the sum of the stress and the stress-rate fields to the sum of the strain and the strain-rate fields in the medium. This transformation defines the functional forms of all components in the symmetric fourth-rank matter tensors that specify the stiffness and the compliance of the dissipative medium. Corresponding stiffness and compliance moduli (e.g., shear modulus and shear compliance) are found to have the same kind of functional dependence upon the frequency. Thus the stress-relaxation effects exhibited by the any stiffness modulus are consistent with the creep (strain-relaxation) effects exhibited by the corresponding compliance modulus. However, as a consequence of the tensor relation between stress and strain, stiffness (and compliance) moduli of different kinds (e.g., shear modulus and bulk modulus) vary with frequency in different ways. The matrix forms of the stiffness and compliance tensors are discussed in terms of analogous equivalent networks that can be used to represent the mechanical behavior of the viscoelastic medium. Tensor transformations and network-analysis algebra were carried out using computer programs for symbol manipulation.

10:30

BB9. An experimental study of the polar characteristics of the group and phase velocities of Lamb waves in graphite/epoxy composite plates. Wade R. Rose, S. I. Rokhlin, and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

The anisotropy of graphite/epoxy composite plates was studied using measurements of phase and group velocities of Lamb waves. Three types of composite structures were used: unidirectional, two directional with orthogonal fibers, and quasiisotropic. The phase and group velocities and the angle of deviation between them were measured using angle variable contact ultrasonic transducers which were situated on the surface of the plates. The anisotropy of the plates resulted in the deviation of the ultrasonic beam from the wave normal which was in some cases more than 60°. This beam deviation had to be taken into account to correctly measure phase and group velocities. The experimental data were compared with calculations of the angular dependence of the phase and group velocities of bulk waves in the graphite/epoxy composite material. The advantage of the Lamb-wave technique over bulk waves is the ability to measure the in-plane anisotropic properties of thin composite plates such as those used in actual aircraft applications.

10:45

BB10. Rigid, acoustically transparent plastic based on fluoroepoxy. Corley M. Thompson (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337) and James R. Griffith (Naval Research Laboratory, Chemistry Division, Washington, DC 20375-5000)

The performance of many underwater acoustical systems would be greatly improved by the use of a structurally rigid material that is nearly transparent acoustically. This means that the material possesses sound speed and density each equal to that of the water medium. The best available commercial materials show a sound speed that is 40% higher than water, and thus are not quite nontransparent acoustically. In earlier work [C. M. Thompson and J. R. Griffith, J. Acoust. Soc. Am. Suppl. 1 70, S74 (1981)] it was shown that a new material based on a partially fluorinated epoxy has much promise. This presentation reports the optimization of the fluoroepoxy by compounding with two different microballoon fillers. The density and sound speed are reported with this choice and are shown to be independently adjustable. Thus a precise match of each with the medium is possible. Samples were fabricated and measured for acoustic transparency by use of panel tests. Results at frequencies down to 10 kHz are presented. The nonacoustic properties are described, and the identity of a commerical source of the material are presented.

11:00

BB11. Influence of shear modulus on the behavior of acoustically transparent materials. Pieter S. Dubbelday and Kurt W. Rittenmyer (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337)

Materials are under development that match density (ρ) and dilatational sound speed (c) to the values of seawater as closely as possible while maintaining sufficient rigidity to serve for structural purposes. Another paper in this meeting by C. M. Thompson and J. R. Griffith (preceding abstract) describes the composition of the material and some of its properties. Matching of density and speed results in transparency for fluids only; the shear modulus in a solid admits the presence of a shear wave which causes deviation from ideal ρ-c behavior. In this paper, the effect of a finite shear modulus on the reflection of plane waves by an infinite plane is analyzed. The shear modulus of the material was measured following a method developed by R. L. Adkins in 1966. Examples are given of the reflection coefficient as a function of incidence angle for values of ρ and c close to those of the medium, and various ratios of plate thickness to dilatational wavelength. The effect of a finite loss tangent in the shear modulus is shown. [Work supported by the Office of Naval Research.]

11:15

BB12. Dynamic bulk moduli of several elastomers. Jay Burns (Florida Institute of Technology, Melbourne, FL 33901), Pieter S. Dubbelday, and Robert Y. Ting [Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337)

Dynamic bulk moduli are given as functions of frequency for several types of elastomers over a temperature range of $-10\,^{\circ}\mathrm{C}$ to $+40\,^{\circ}\mathrm{C}$. Measurements were made by a modification of the acoustic coupler method of McKinney *et al.* [J. Appl. Phys. 27, 425 (1956)]. Values of K', the real or storage part of the complex modulus, fell into the range 2 to 4 GPa, typical of values for organic liquids. The loss tangents fell into the range below 5% and were mostly less than 2%, also typical of many liquids. This similarity to liquids suggests that similar short-range dynamic mechanisms dominate the compressional free-volume models. One neoprene sample exhibited the full transition from low to high limiting values of K'' over the experimental temperature range, thereby providing a good test for the applicability of WLF-type temperature "shifting" in these materials. A master curve was constructed for this sample. It gave evidence for a relatively broad spectrum of relaxation times.

11:30

BB13. Material measurements of two independent viscoelastic coefficients using the Metravib-04 viscoanalyzer. R. G. Kasper and A. B. Bruno (Research and Technology Staff, Naval Underwater Systems Center, New London, CT 06320)

A number of dynamic mechanical measurements have been completed on the Metravib-04 viscoanalyzer for a variety of solid materials. The materials tested were elastomers, foams, and metal/elastomeric composites. For each batch, the materials were tested in the tension-compression mode and in the shear mode in a frequency range of 5.0 Hz to 1.0 kHz at constant temperature. The data produced two independent complex elastic modulii, Young's modulus and shear modulus, as a function of frequency at room temperature. For assumed isotropic and homogeneous conditions, the dilatational and shear-wave speeds were independently calculated for each material. The imposition of the material incompressibility condition (i.e., Poisson's ratio equal to 0.5) was not required. These values were used as input functions for numerical models of transducer isolation components subjected to various harmonic excitations. A number of examples will be discussed.

11:45

BB14. Spurious resonances in adaptation of elastic layer model to simulate liquid layers. Jacob George (Mail Stop 181, Raytheon Submarine Signal Division, Portsmouth, RI 02871)

When computer programs based on an elastic layer model [e.g., Folds and Loggins, J. Acoust. Soc. Am. 62, 1102 (1977)] are used to simulate liquid layers, spurious resonances appear as a function of frequency. It is shown that they are not caused by numerical precision limitations in the computer, but are inherent in the model. In the computer representation of the elastic layer model, the limit of Lamé constant $\mu \to 0$ is approximated by assigning a very small value for μ. With such a choice, spurious resonances appear when transmission coefficient is plotted as a function of frequency. This is also proven analytically, and the mechanism of typical resonances with graphs is illustrated. For frequencies away from spurious resonances, the model does yield the correct liquid layer results.

12:00

BB15. Anisotropic wave propagation in crystals: Quasilongitudinal and quasitransverse waves. G. Kameswara Rao (Department of Physics, Chaitanya Bharathi Institute of Technology, Hyderabad-500 171, India)

We can propagate elastic waves in any crystal plane along the principal directions as well as along certain specific directions as purely longitudinal and purely transverse waves. [F. E. Borgnis, Phys. Rev. 98, 1000–1005 (1955)]. In some cases, the waves deviate from their pure nature and such waves are called quasilongitudinal and quasitransverse waves, in which the displacement direction does not exactly coincide with the propagation direction. In the case of orthorhombic crystals, for a wave propagating in the {110}, planes an expression for the deviation angle 0 is derived in terms of the elastic constants and the direction cosines of the propagation vector. The deviation angles have been calculated for 11 crystals. In the case of orthorhombic crystals, the lattice parameters are designated with the convention $b > a > c$. [W. R. Cook, Jr. and H. Jaffe, Acta Cryst. 10, 705 (1957)]. But a survey of the crystallographic data shows that this has not been so. Only in the case of 4 crystals out of the 11, does the crystallographic data fit the $b > a > c$ convention. This necessitates the reconsideration of the elastic constants data for the purpose of rationalization. A knowledge of the deviation angles enables one to assess the order of accuracy of the experimentally determined elastic constants of the type c_{ij} $(i \neq j)$. The order of accuracy of the c_{ij} $(i \neq j)$ is of particular importance in calculating the contribution due to stress birefringence or stress optical path retardation in the determination of the photoelastic and electro-optic constants of the crystals.

THURSDAY MORNING, 15 MAY 1986 WEST BALLROOM, 8:15 A.M. TO 12:00 NOON

Session CC: Physiological Acoustics V, Psychological Acoustics VI, and Musical Acoustics I: The Psychology and Physiology of Pitch Perception and Frequency Discrimination

William M. Hartmann, Chairman
Physics Department, Michigan State University, East Lansing, Michigan 48824

Chairman's Introduction—8:15

Invited Papers

8:20

CC1. Pitch perception in impaired listeners. Edward M. Burns (Department of Speech and Hearing Sciences, JG-15, University of Washington, Seattle, WA 98195)

The study of pitch perception in hearing-impaired listeners is of interest primarily because of the hope that measurements of deficits in pitch perception in listeners with particular types of impairments, in conjunction with other psychoacoustic and physiologic measurements of these same impairments, will lead to insights into the process by which pitch information is encoded in the normal system. This paper reviews studies of frequency discrimination, and pure- and complex-tone pitch perception, in impaired listeners. Among the impairment-related phenomena which will be discussed are: (1) the degradation of frequency discrimination and increase in pitch-matching variability; (2) the exaggeration of various pitch-shift phenomena such as pitch-level effects, diplacusis, musical paracusis, and post-stimulatory pitch shifts; and (3) the degradation of complex tone pitch perception. In addition, other anomalous pitch-related phenomena, which may or may not be considered impairment related, such as tonal tinnitus, tonal monaural diplacusis, and polyacusis, will be reviewed. Finally, the perception of pitch-like sensations by cochlear implantees will be examined.

8:55

CC2. Modeling pitch perception of complex tones. Adrianus J. M. Houtsma (Institute for Perception Research, P. O. Box 513, 5600 MB Eindhoven, The Netherlands)

When one listens to a series of harmonic complex tones that have no acoustic energy at their fundamental frequencies, one usually still hears a melody that corresponds to those missing fundamentals. Since it has become evident some two decades ago that neither Helmholtz's difference tone theory nor Schouten's residue theory could adequately account for this phenomenon, several other theories have been proposed that accentuate central neural rather than peripheral mechanical signal processing. Some of these theories will be critically reviewed against empirical evidence from recent psychoacoustic studies. In particular, the relative advantages and disadvantages of the descriptive "virtual pitch" theory of Terhardt and the stochastic "optimum processing" theory of Goldstein are discussed in relation with recent data on pitch perception for simultaneous complex tones. Taped examples of some of the studied phenomena are provided.

9:30

CC3. Criteria and constraints of modeling the perception of pitch. Ernst Terhardt (Institute for Electroacoustics, Technical University, Arcisstr. 21, D-8000 Munich 2, West Germany)

The vast amount of data and observations relevant to the perception of pitch has grown through many decades and at first did not seem to support ease of modeling; it rather elucidated that pitch modeling is much more complex than just measuring "the fundamental frequency" in either the frequency or time domain. While the concept of auditory spectrum analysis is enormously helpful and successful in many respects, it failed to provide a direct explanation of the pitch of many ordinary sounds such as speech and music. Various time-domain explanations, on the other hand, proved unsatisfactory as well. In the late 1960s the conclusion was drawn that modeling pitch perception requires careful inclusion, and combination, of signal-theoretical, psychophysical, and *Gestalt*-psychological methods. The present author's "virtual-pitch theory" was worked out on that line. As an additional significant outcome of the new approach, it turned out to be not quite just a pitch model, but reflects and illustrates significant general principles of auditory perception and, in particular, of music perception. Until present, that type of modeling has been successful and appears to be supported by more recent experimental findings. Thus critical reconsideration and analysis of its *essentials*, i.e., indispensable characteristics, structures, and parameters seems appropriate. Particular attention is paid to the characteristics of aural spectrum analysis, the dichotomy of spectral *versus* virtual pitch, and the question of learning processes. Some crucial phenomena are demonstrated using tape examples.

10:05–10:15

Break

10:15

CC4. Physiological observations on the neural representation of complex pitch. E. F. Evans (Department of Communication and Neuroscience, University of Keele, Keele, Staffordshire ST5 5BG, United Kingdom)

Pitch models which depend on the manner in which the frequencies of pitch-evoking stimuli are encoded in the temporal discharge patterns of neurones in the auditory periphery, would be easier to test if it were possible to predict the temporal discharge patterns in response to such complex stimuli. As an extreme case, the effects of variants of the click trains, whose pitch has been the subject of controversy between Whitfield and Moore (see Evan's chapter in *Hearing—Physiological Bases and Psychophysics*, edited by R. Klinke and R. Hartmann (Springer, New York, 1985) have been studied at the level of the cat's cochlear nerve. In particular, the following stimuli have been used on fibers with characteristic frequencies up to 4 kHz: pulse trains with even intervals of 5, 10, 20 ms; uneven intervals of 4.5/5.5 ms, 4.6/5.4 ms, 4.8/5.2 ms, and even intervals of 2.5 and 5 ms with alternate click phases. Autocorrelations have been performed of the spike discharge patterns evoked by the stimuli. To a first approximation, temporal patterns of discharges evoked by the even and uneven interval click trains can be predicted on the basis of simple linear filtering, with filters having the shape and bandwidth characteristics of the fiber's frequency threshold (tuning) curve (FTC). A simple probabilistic model of a cochlear fiber [E. F. Evans, J. Physiol. **298**, 6–7 (1980)] preceded by a linear filter having these characteristics predicts the observed patterns well. In particular, the temporal patterns are determined by the degree to which individual harmonics of a complex stimulus interact, after appropriate weighting by the cochlear filter. Two important consequences of this follow. One is that the dominant region for pitch could be considered to be the range of cochlear fiber characteristic frequencies over which sufficient spectral resolution

for place models coexists with sufficient interaction of spectral components as required by pitch models based on the microstructure of the temporal pattern of spike discharges. The second is that spectral filtering needs to be incorporated into any temporal fine-structure model: One result is the observed relative insensitivity of the temporal patterns to stimulus component phase, contrary to the objection frequently raised against temporal fine-structure models.

10:50

CC5. Pitch and the perceptual organization of complex spectra. W. M. Hartmann (Physics Department, Michigan State University, East Lansing, MI 48824)

The psychological attribute of pitch is the most important organizing element by which the human auditory system reduces a steady-state complex signal to one or more perceptual entities. Historically both experimental and theoretical studies of pitch perception have been primarily concerned with the synthetic listening mode and the pitch of a single entity. However, there are experiments which test the analytic mode and the ability to segregate several entities. The mistuned harmonic experiment, in which the task is to hear out a mistuned partial in a complex tone, is a promising approach of this kind, though it demands that the experimenter try to distinguish those effects which are due to beats of mistuned consonances or nonlinear combination tones from those due to perceptual reorganization into segregated entities. The results of mistuned harmonic experiments suggest that the most sensitive segregation operation is mediated by neural processes which are tonotopically local, though they are tuned more broadly than a critical band. Therefore, appropriate models are not based upon pattern matching in a central spectrum but upon cross correlation between the firing patterns in neighboring sharply tuned neural elements or upon autocorrelation of the pattern in a more broadly tuned element. Such models are supported and constrained by physiological observations of the temporal response to complex tones in single cells of the eighth nerve, the cochlear nucleus, the inferior colliculus and the cortex.

11:25

CC6. Complex pitch—Effects of context and experience. Joseph W. Hall, III (Department of Communication Sciences and Disorders, Northwestern University, 303 E. Chicago Avenue, Chicago, IL 60611), Robert W. Peters (Division of Speech and Hearing Sciences, University of North Carolina at Chapel Hill, Chapel Hill, NC 27514), and David R. Soderquist (Department of Psychology, University of North Carolina at Greensboro, Greensboro, NC 27412)

Complex pitch perception is influcenced not only by the physical composition of the stimulus, but also by the context in which the stimulus occurs, and by previous pitch stimuli to which the listener has been exposed. Three complex pitch effects will be discussed: (1) Pitch for nonsimultaneous harmonics presented in quiet and in a noise background; (2) short-term complex pitch adaptation effects; (3) long-term changes in complex pitch perception where the pitch associated with one complex tone changes to the pitch associated with a second complex, after the two complex tones are paired and presented to the listener over several hours of listening. Findings will be related to pitch theories, plasticity of pitch perception, and possible organization of complex pitch "channels."

This session was sponsored in part by a grant from the National Institutes of Health to Michigan State University.

THURSDAY MORNING, 15 MAY 1986 EAST BALLROOM, 8:30 TO 10:59 A.M.

Session DD: Speech Communication VI: Human and Machine Perception

Thomas H. Crystal, Chairman
Institute for Defense Analysis (CRD), Princeton, New Jersey 08540-3699

Chairman's Introduction—8:30

Contributed Papers

8:35

DD1. Implications of research with nonspeech stimuli for speech processing: A summary of recent studies using world-length tonal patterns. C. S. Watson (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

A series of discrimination experiments with word-length sequences of brief tones has demonstrated aspects of auditory processing not evident from studies with simpler stimuli (single tones, noise bursts, clicks). The primary new limitations on processing discovered with the nonspeech stimulus patterns are associated with (a) the level of stimulus uncertainty (size of the catalog sampled to generate a sequence of trials) and (b) the information content within the individual patterns (represented by the number of freely varying components per pattern). These jointly determine the level of *informational masking* within the patterns, a central limit on component detectability that can vary from 0 to 40–50 dB above the expected (peripherally determined) intercomponent temporal masking. These empirical results are interpreted in terms of three properties of auditory memory: long-term memory (LTM), the accessibility of the properties of individual complex waveforms within the LTM, and the informational capacity of short-term auditory memory. Informational masking is reduced or eliminated under conditions that permit LTM-based (top-down) selective attention to meaningful portions of a complex nonspeech waveform. It is proposed that precise discrimination of critical elements of speech waveforms depends on similar central-control mechanisms. [Supported by NINCDS and AFOSR.]

8:47

DD2. Informational masking in vowel sequences. D. Kewley-Port, C. S. Watson, and M. Czerwinski (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

In a number of earlier studies, Watson and his colleagues have described various stimulus and training variables which determine the detectability and discriminability of single components within complex sequences of tones [e.g., J. Acoust. Soc. Am. **76**, 1037–1044 (1984)]. A series of studies has begun to investigate whether, under the same conditions, similar discriminability will be observed for speech and nonspeech waveforms. Modeling a female talker, ten English vowels were synthesized at five durations (20–160 ms). Detectability of the vowels was studied using an adaptive tracking task. Large (20 dB) differences in thresholds were obtained among different vowels presented in isolation. Thresholds were then obtained for 40-ms vowels presented in ten-vowel sequences. In a high-uncertainty task, parameters previously shown to affect detection in tonal sequences similarly affected vowel detection. Vowels in initial or final position were detected most easily. Informational masking for vowels was observed to vary from 0 to 45 dB across subjects. At least for these preliminary experiments, the same mechanisms shown to underlie the perception of tonal sequences also appear to determine the processing of complex speech stimuli. [Supported by NINCDS and AFOSR.]

8:59

DD3. Spectral envelopes and perceptual target zones for consonants and vowels: Preliminary estimates. James D. Miller and John W. Hawks (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

Preliminary estimates of the sizes, shapes, and locations of perceptual target zones (PTZ) of ten vowels, six dipthongs, a twenty-nine consonants have been made. The PTZ's are defined as enclosed volumes in the space $x = \log(PF3/PF2)$, $y = \log(PF1/PR)$, and $z = \log(PF2/PF1)$, where PR is the perceptual reference in Hertz and PF1, PF2, and PF3 are the center frequencies of the perceptual formants. The perceptual formants are estimated from spectral envelopes as guided by rules for distinguishing burst-friction spectra and glottal-source spectra and for naming and numbering formants based on their logarithmic relations to the value of the reference. The estimation of PTZ's will be illustrated for at least one allophone from each of the following classes: vowels, diphthongs, approximates, nasals, voiceless fricatives, and voiced fricatives. These preliminary estimates indicate that PTZ's: (1) are large, thus encompassing a considerable range of spectral variants as might be associated with coarticulatory effects and articulatory style; (2) have irregularly shaped, abutting boundaries; and (3) for the most part, exhibit little overlap. [NINCDS grant R01-NS21994-01.]

9:11

DD4. Interaural contrast effects in vowel perception. Robert Allen Fox (Speech and Hearing Science, Department of Communication, The Ohio State University, 324 Derby Hall, 154 North Oval Mall, Columbus, OH 43210)

Many recent experiments have suggested that the contrast effect found in vowel identification is not a function of response bias or detector fatigue, but rather of perceptual processes such as adaptation level. However, it is unclear as to whether these contrast effects are both peripheral and central in nature and whether or not subjects must attend to (and/or identify) the contrasting elements. In the present anchoring experiment, subjects identified vowels from a *hid–head* continuum under two conditions: an equiprobable control condition and an anchoring condition. In the control condition, the stimuli were only heard in one ear, but there were three different anchoring conditions: (1) both targets and anchors occurred in one ear only; (2) targets occurred in one ear while anchors occurred in the opposite ear—all tokens were identified; (3) targets occurred in one ear while anchors occurred in the opposite ear—only targets were identified. As expected, when all stimuli were heard in the same ear, anchoring produced a significant shift in the *hid–head* phoneme boundary. However, when achors occurred in the opposite ear, a significant shift was obtained only when the anchor was identified. It will be suggested that these results indicate that the vowel contrast is central in nature and may require the focus of attention-driven mechanisms.

9:23

DD5. The effects of similarity neighborhood structure on auditory word recognition. Paul A. Luce (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

The present study examined how the structure of similarity neighborhoods of spoken words affects auditory word recognition. Similarity neighborhoods were computed on phonetic transcriptions of approximately 20 000 words in an on-line version of Webster's pocket dictionary. Among the variables of interest were: (1) the density of the similarity neighborhood; (2) the frequencies of words within the neighborhood; and (3) the frequency of the target word itself. Nine-hundred and twenty-eight three-phoneme, monosyllabic words varying on each of these variables were presented at each of three signal-to-noise ratios for identification. In addition, confusion matrices were obtained for all initial and final consonants and vowels occurring in the 928 stimuli in order to account for basic differences in segmental confusability and intelligibility. The results will be discussed in terms of the combined effects of similarity neighborhood structure and segmental confusability on auditory word recognition. [Work supported by NIH grant No. NS-12179 to Indiana University.]

9:35

DD6. Speech in noise revisited: Evaluation of intelligibility and voice quality of speech in broadband noise. Caldwell P. Smith (Electromagnetic Sciences Division, Rome Air Development Center, RADC/EEV, Hanscom AFB, MA 01731)

Recordings of diagnostic rhyme tests (DRT) for intelligibility and sentence lists for quality assessment spoken by three male speakers were mixed with white noise and low-pass filtered at 4 kHz. Calibration of speech-to-noise energy ratios was accomplished with the Sims algorithm [J. T. Sims, J. Acoust. Soc. Am. **78**, 1671–1674 (1985)] to obtain S/N ratios of 6, 12, 18, 24, and 30 dB. The resulting composite signals were recorded with high-quality digital audio equipment (Sony PCM-F1) which was also used to reproduce the signals for presentation to listeners. Voice quality and acceptability were assessed with the diagnostic acceptability measure (DAM) test of Voiers, and the mean opinion score (MOS) rating. Data from analysis of listener responses were used in calculating regression model relating scores for overall intelligibility, feature intelligibility, signal quality, background quality, overall quality, and mean opinion scores, to S/N energy ratios. Models fitted to overall intelligibility scores, as well as some of the intelligibility feature scores, were significantly improved with regard to values of r^2 (multiple correlation coefficient squared) by using indicator variables for speakers, i.e., regression models with separate origins for speakers, but a common slope for the regression lines. Regression models also facilitated interpolation of scores to compare descriptive categories that have been used for intelligibility, quality, and MOS scales; e.g., "fair" has been used to categorize intelligibility between 79 and 83, quality scores between 42 and 48, and MOS ratings between 2 and 3. For speech in broadband noise, categories varied widely for the three scales.

9:47

DD7. Perception and interpretation of English intonation by Arabs. Fares Mitleb (Department of English, Yarmouk University, Irbid, Jordan)

This paper outlines an experiment designed to assess Arab's perception and interpretation of English intonation. Subjects were 30 native Arabic speakers majoring in English. Test material was a set of 20 minimally paired sentences, differing in intonation only, or presenting the same intonation pattern twice, spoken on a tape by a native English speaker. The task was to decide whether the two sentences in each pair have the "same" or "different" intonation pattern and to assign meaning(s) of the provided glosses according to the previous judgement. The analysis of the responses indicates a clear hesitation in perceiving intonation and in assigning meaning to intonation patterns tested in this study. This hesitation, however, is not attributed to intonation interference; rather, it seems to be a result of faulty instructional process. English pronunciation is usually taught to second language learners with great emphasis on the segmental language units. Language learners usually preserve this knowledge when they are moved into suprasegmental situations. We thus propose that English pronunciation is to be introduced in suprasegmental units instead of segmental units.

9:59

DD8. Automatic target generation for vowels. Hisao M. Chang and James D. Miller (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

Research in phonetic recognition by computer is of interest because it may offer better performance than systems using larger-recognition units such as syllables or words. Among several techniques developed to model speech signals as phoneme sequences (e.g., hidden Markov models and other feature-based methods) is the phonetic-target model based on the auditory-perceptual theory of Miller [ASHA Reports 14 (1984)]. In it: (1) talker differences that mask acoustic phonetic correlates are minimized when incoming speech signals are represented as paths in a three-dimensional space, where $X = \log(P3/P2)$, $Y = \log(P1/R)$, and $Z = \log(P2/P1)$; and (2) phonetic targets defined in this space seem to be uniquely related to the acoustic patterns of phonelike elements. An automatic procedure generated three phonetic targets for the vowels /ɪ/, /ɛ/, and /ʌ/ from a set of training tokens in stop–vowel–stop format recorded from two male and two female talkers. When four new talkers were tested, a vowel recognition accuracy of about 98% was attained. Additionally, a segmentation algorithm isolated the vocalic segments from the stop bursts with 100% accuracy. [NINCDS grant R01-NS21994-01.]

10:11

DD9. Toward a massively parallel system for word recognition. Maurice K. Wong (GTE Laboratories Incorporated, 40 Sylvan Road, Waltham, MA 02254) and Hon Wai Chun (Computer Science Department, Brandeis University, Waltham, MA 02254)

This paper describes a massively parallel system for word recognition. Based on the connectionist network model, the system consists of a large number of simple neuronlike processing units, or nodes, which represent words, phonetic segments, or phonetic features. The computation consists of constant updating of activation levels of all nodes, resulting from the excitatory links and inhibitory links between the nodes. Input to the system consists of frame-by-frame scores of similarity to a set of predefined spectral filters, which represents the set of phonetic segments necessary for distinguishing between words in the vocabulary. These similarity scores are combined into phonetic feature indexes for each frame of speech as input to the feature nodes in the network. A linguistic knowledge base is built into the network, allowing both data-driven processing and top-down prediction to cooperate or compete in working toward the correct lexical hypothesis. The system has been implemented using a software package simulating massively parallel networks on a lisp machine. Preliminary testing of the system using digits and nine letters by one speaker is 100% successful in spite of the low frame-by-frame recognition accuracy.

10:23

DD10. Automatic analysis of speech using parallel cellular pipelined processor. John F. Hemdal (Department of Electrical Engineering, The University of Toledo, Toledo, Ohio 43606) and Robert H. Lougheed (Environmental Research Institute of Michigan, Ann Arbor, MI 48107)

Early results of analyzing natural speech using a parallel processor were reported before this society in Minneapolis in the fall of 1984. This paper is a presentation of subsequent results obtained. The CYTO-HSS™ parallel pipelined processor, used for analyzing speech spectrograms, is a work station for image analysis, optimized for cellular automata algorithms. The HSS has several advantages over typical Von Neumann type machines. One advantage is speed. In performing similar image processing tasks, the CYTO-HSS™ is from 100 to 10 000 times faster than the VAX. The distinctive features of speech are described in mathematical morphological terms and the parallel processor detects the presence of the features. Examples of stop and fricative detection will be described and shown. Potential applications in speech analysis, automatic speech recognition, speech synthesis, bandwidth compression, and speech enhancement are discussed.

10:35

DD11. A method for automatic phoneme boundary detection. Hany Selim (IBM Cairo Scientific Center, 56 Gameat El Dowal El Arabia Street, Mohandessin, Cairo, Egypt)

A three-way classifier is introduced, which reliably differentiates between voiced, unvoiced, and silence segments of a speech utterance. The extreme points of the median smoothed first difference of the output of this classifier are used for the phoneme boundary detection. The classification criterion is calculated by weighting the log of the ratio of high- and low-pass filtered versions of the speech utterance by clipped versions of a normalized root average energy and a normalized zero crossing rate. The high- and low-pass filtering operations were simply performed by calculating the sample difference signal and the sample addition signal. The frequency domain attenuation characteristics corresponding to these simple filters are those of a quarter period of a cosine and sine waves, respectively. These filters were found to be completely sufficient for the purpose of classification. The phoneme boundary detector can be used for the mechanized acquisition of the phoneme inventory of any language and, due to the simple operations involved, in automatic speech recognition.

10:47

DD12. Negative binomial probability models for speech intelligibility data. Caldwell P. Smith (Electromagnetic Sciences Division, Rome Air Development Center, RADC/EEV, Hanscom AFB, MA 01731)

Since reporting at ASA•50 on success in fitting negative binomial probability distributions to speech intelligibility data, extensive additional analyses have been made on Diagnostic Rhyme Test intelligibility data from multi-speaker tests. Modeling involves converting feature scores to frequency counts of listener errors, for each speaker's scores. Scores for the four primary feature states for each of the six phonetic attributes of the DRT represent outcomes of presenting eight tokens to eight listeners; thus error frequency counts from zero to 64 are possible, for each of the 24 attribute states. After conversion of scores to frequencies, a cumulative distribution was formed from the ranked data, a "frequency of frequencies" distribution. The mean and variance of these distributions provided parameters for negative binomial distributions, which were compared with chi-squared and Kolmogorov–Smirnov tests. Similar tests were performed on sub-sets of the intelligibility data representing partitioning into scores for individual speakers, for all male speakers, all female speakers, voiced feature scores, and unvoiced feature scores. Analysis of intelligibility data from tests of a variety of conditions revealed that in the majority of cases these distributions did not differ significantly from the negative binomial probability models, either for the composite data or the partitioned data sets. These findings open up new avenues for assessment and interpretations of speech intelligibility data.

Session EE: Underwater Acoustics V: Rough Surface Scattering and Multipath Propagation

Alexandra Tolstoy, Chairman
Naval Research Laboratory, Washington, DC 20375

Chairman's Introduction—8:30

Contributed Papers

8:35

EE1. Inversion techniques for obtaining seabed low-frequency reflection loss at small grazing angle in shallow water. Ji-xun Zhou[a] (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Low-frequency bottom reflection loss at small grazing angles is a key parameter for sound field prediction in shallow water, but it is difficult to measure directly. For small grazing angle, the relationship between the bottom reflection loss and the grazing angle is nearly linear, $-20\log|V(\theta)| = B\theta$, where $B \propto K_p f^{(n-1)}$ and sound attenuation in the sediments is expressed by $\alpha_p = K_p f^n$ (dB/m·kHz). The bottom reflection loss parameter B in shallow water can be extracted by inversion techniques from a number of different field characteristics. In this paper, some theoretical inversion methods and extract reflection-loss parameters (B) for different sea areas are presented. The experimental results show that the parameter B apparently increases with increasing frequency. This implies that $n > 1$, i.e., that the frequency dependence of the sound attenuation in the sediments could be nonlinear. The experimental results of E. C. Lo *et al.* [J. Acoust. Soc. Am. **74**, 1833–1836 (1983)] can be explained very well by using the frequency dependence of bottom attenuation (and sound velocity) obtained by inversion of the normal mode measurements made by J. X. Zhou [J. Acoust. Soc. Am. **78**, 1003–1009 (1985)]. [a] J. X. Zhou on leave (1985) from Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China.

8:50

EE2. The second Born approximation and high-frequency scattering at low grazing angles. Stanley A. Chin-Bing, Michael F. Werby, and Steve Stanic (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

The first Born approximation has proven to be a successful tool to describe scattering from impenetrable interfaces at high frequency providing the existence of secondary scattering from adjacent parts of the surface is unimportant. This usually means that low grazing angle scattering is avoided for a suitably rough surface where secondary scattering becomes important. To treat secondary scattering from adjacent parts of the interface, one must resort to higher-order Born terms. In particular, the second Born term can account for one secondary event which becomes important at a suitably low grazing angle (depending on surface roughness). In this study, a formulation based on the second Born treatment for scattering is presented and the results are compared with the conventional first Born contribution.

9:05

EE3. Scattering loss of interface waves at a rough seabed: Theoretical calculations. Pierre Zakarauskas, David M. F. Chapman (Defense Research Establishment Atlantic, P. O. Box 1012, Dartmouth, Nova Scotia, Canada, B2Y 3Z7), and Anthony J. Purcell (6266 London Street, Halifax, Nova Scotia, Canada B3L 1X1)

Using the Rayleigh method, a boundary perturbation technique, the propagation of interface waves (i.e., Scholte waves) along a randomly rough fluid/solid boundary is studied. It is assumed that the surface has Gaussian statistics and that the autocorrelation function is $\exp(-r^2/L^2)$, in which L is the correlation length. The method assumes that the rms surface height is small compared to the acoustic wavelength λ and that the surface slopes are also small. There is no restriction on L/λ. The intensity of the average field decays exponentially with range; the energy lost is scattered into interface waves traveling in other directions and into bulk waves carrying energy away from the boundary. For $L/\lambda \ll 1$, the loss varies as $(L/\lambda)^5$ and energy is lost mostly to bulk waves; for $L/\lambda \gg 1$, the loss varies as $(L/\lambda)^4$ and the energy is lost mostly to interface waves; at intermediate values of L/λ, there is a local maximum of energy loss. This behavior is explained qualitatively using wavenumber-space scattering diagrams. Results for the cases of water over sand, chalk, and granite are presented, respectively.

9:20

EE4. A further investigation of acoustic scattering from the sea surface. Bernd Nuetzel, Heinz Herwig (Forschungsanstalt der Bundeswehr fuer Wasserschall-und Geophysik, 23 Kiel 14, Federal Republic of Germany), Joseph M. Monti, and Paul D. Koenigs (Naval Underwater Systems Center, New London, CT 06320)

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-beamwidth) pulsed parametric sonar transmitter and conventional receivers. Scattering strength values were obtained as a function of frequency (3–18 kHz) and grazing angle (10°–90°) for differing sea surface roughness and wind speed conditions. The mean scattering strength values illustrate a strong dependence on large and small scale sea surface roughness. Comparisons with theoretical and semiempirical models indicate the appropriateness of using composite roughness modeling theories.

9:35

EE5. Inversion of data for near-grazing propagation over rough surfaces. R. J. Lucas[a] and V. Twersky (Mathematics Department, University of Illinois, Chicago, IL 60680)

We apply our results for the coherent response to a point source irradiating an embossed plane [J. Acoust. Soc. Am. **76**, 1847 (1984)] to interpret and invert data obtained by Medwin *et al.* [J. Acoust. Soc. Am. **76**, 1774 (1984)]. As emphasized earlier [J. Acoust. Soc. Am. Suppl. 1 **74**, S122 (1983)], key features arise from interference of elementary components of the corresponding Sommerfeld type wave system based on the complex error function complement Q. Simplified analytical approximations are presented to emphasize the essential physics, and to facilitate initial estimates of unknown parameters. Refined estimates using Q integral theory are incorporated in graphical results to display primary data trends of the magnitudes and phases as well as the dependence of the incremental dispersion on both range and frequency. [a] Visiting from Department of Mathematical Sciences, Loyola University, Chicago, IL.

9:50

EE6. Angular dependence of under ice reflectivity in the marginal ice zone (MIZ). Patricia L. Gruber and Ronald L. Dicus (Code 5120, Naval Research Laboratory, Washington, DC 20375)

Acoustic reflectivity measurements from sea ice in the MIZ were taken during MIZEX-84. Signals from explosive charges (MK-61 and -82 SUS) received on 12 hydrophones of a 313-m vertical array were processed to separate single bounce ice reflections from direct path and bottom reflected arrivals by replica deconvolution. Ice reflection loss was computed from the ratio of reflected energy to direct path energy after correction for spreading loss differences between the two paths. Signals from different source ranges and receiver depths were collected on the basis of their specular reflection grazing angles spanning the interval 10° to 30°. Data points with estimated standard deviation of reflectivity greater than 0.2 (based on noise estimates) were excluded. The resulting reflectivity versus grazing angle curves at 64, 96, and 128 Hz displayed high variance presumably due to scattering from a single under ice pressure ridge within the Fresnel zone. With angular smoothing the curves showed small angular dependence and losses on the order of 1.5 dB. Comparison of results with model computations will be discussed.

10:05

EE7. A stochastic propagation and scattering operator for ocean propagation. C. C. Yang (Department of Electrical Engineering, Pennsylvania State University, University Park, PA 16802) and Suzanne T. McDaniel (Applied Research Laboratory, Pennsylvania State University, University Park, PA 16802)

General properties of acoustic propagation and scattering in a stochastic open environment are investigated. In this study, the positions of the source and observation points can be arbitrary. Furthermore, these two points can be in relative motion. The most important factor of the ocean environment affecting wave propagation is the sound-speed profile. Also, because of the fine structure in temperature, stochastic fluctuations of the sound speed may exist along the paths of a sound wave. Therefore, effects from random scattering have to be taken into account. Because of the existence of the sound-speed profile in the ocean, more than one path may be drawn to connect the source and observation points. This multipath effect, including that due to the correlations among these paths, is fully discussed. To make computations feasible, the path integral technique is applied. [Work supported by ONR.]

10:20

EE8. The effect of an El Niño/Southern Oscillation event on underwater sound propagation. D. R. Palmer, L. M. Lawson, Y.-H. Daneshzadeh, and D. W. Behringer (National Oceanic and Atmospheric Administration, Atlantic Oceanographic and Meteorological Laboratory, 4301 Rickenbacker Causeway, Miami, FL 33149)

From June 1981 through June 1984, 15 separate ocean cruises were made to the equatorial Pacific to collect hydrographic data at 85°W as part of a NOAA climate program. The most recent El Niño/Southern Oscillation (ENSO) event occurred during this period. Temperature and salinity data obtained during the cruises have been processed and merged with archival data to form a series of sound-speed profiles which reflect the onset, evolution, and cessation of the ENSO event. By using computer models, with these profiles as input data, it is shown that propagation conditions fundamentally changed as a result to the ENSO event. Conditions which support a convergence-zone structure for a shallow source disappeared during the event resulting in bottom-limited propagation. While this severe change provides an opportunity to detect and monitor ENSO events using underwater sound, it prevents the straightforward application of acoustic tomography to obtain the temperature field and, hence, to study the dynamics of the event. For a source located at middepth, however, tomographic techniques may be applicable since propagation conditions are not as changeable there.

10:35

EE9. An analysis of the temporal fluctuations of cw acoustic propagation in the marginal ice zone. Peter H. Dahl, Arthur B. Baggeroer (MIT/WHOI Joint Program in Oceanography, MIT, Cambridge, MA 02139), and Peter N. Mikhalevsky (Science Applications International, Falls Church, VA 22046)

Continuous wave (cw) acoustic transmission data from MIZEX 84 (marginal ice zone experiments) were transmitted between two ships separated by approximately 100 km and propagated via a partially ice-covered path. The signals were stepped in frequency between 25 and 200 Hz for 1 h. Both vessels were drifting freely which resulted in a Doppler shift in the received multipath signal. The measured Doppler compares favorably with available navigational data. The quadrature demodulated received signal is modeled as a (locally) wide-sense stationary process. Accordingly, we can exploit the Gaussian correlation structure and estimate the moments of the power spectrum using the covariance method. This method is computationally efficient and relatively unexploited for this purpose. We relate the rms spectral width directly to the variance of the phase rate (ν^2). The value of ν^2 is a function of oceanic processes which dynamically perturb the sound field. Estimates of ν^2 from the MIZEX data are compared with similar experiments carried out in the more quiescent Central Arctic, and in midlatitude regions.

10:50

EE10. Low-frequency propagation across an East Greenland ice-edge eddy: Winter conditions. Leonard E. Mellberg (Naval Underwater Systems Center, Newport, RI 02841-5047), Ola M. Johannessen (Geophysical Institute, University of Bergen, Norway, N-5014), Donald N. Connors, George Botseas, and David G. Browning (Naval Underwater Systems Center, Newport, RI 02841-5047)

A series of small cyclonic eddies have been observed along the ice edge adjacent to the East Greenland current in the vicinity of the Fram Strait. These eddies are characterized by a unique spiral surface pattern as pack ice is drawn into the circulation. Melt water from this ice contributes to a complicated temperature-salinity structure. An analysis of low-frequency (50 Hz) propagation is presented using environmental range-dependent acoustic prediction models: parabolic equation (PE) and ray model [GRASS]. The environmental data are for winter conditions based on a longitudinal oceanographic transect. Results are compared to previous analyses of a similar eddy in the Fram Strait under summer conditions and the directional dependence of acoustic modes across the East Greenland current frontal zone. [Work supported by NUSC and ONR.]

11:05

EE11. A depth, range, and time probability distribution of intensity for wave propagation in random media. T. E. Ewart (Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, WA 98105)

The second moment of intensity as a function of depth, range, and time for WPRM can be modeled in terms of a space–time autocorrelation function, a scattering strength parameter γ, and a scaled range X. The higher moments have not been predicted for all γ and X; however, it is generally accepted that the intensity moments are lognormal at small X and exponential at large X. The generalized gamma distribution function (GGDF) [E. W. Stacy, Ann. Math. Stat. **33**, 1187 (1962)] forms a large class that includes the lognormal and the exponential pdf's. It is proposed that the GGDF can model the probability distributions of intensity in forward scattering over wide ranges in γ and X. The temporal intensity fluctuations measured during MATE and depth-range results from Monte-Carlo simulations of WPRM for a medium with a power law autocorrelation function have been used to test the proposition. The "goodness of fit" of the GGDF's fitted to those data sets, and the benefits of using distribution modeling rather than intensity moment will be discussed. The results are convincing; it remains a nontrivial task to test the hypothesis theoretically. [Supported by ONR code 425OA.]

11:20

EE12. Observations of acoustic wave propagation in the Arctic Ocean—Preliminary results from AATE. Terry E. Ewart and S. A. Reynolds (Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, WA 98105)

Preliminary results are presented for the AIWEX acoustic transmission experiment (AATE). Pulsed tones at 2, 4, 8, and 16 kHz were transmitted from a fixed source to three depth cycling receivers for 12 days (spanning a depth of 150 m) and to a fixed receiver 100 m away (for the last 7 days). The plane of the four receivers was perpendicular to the transmission path. Extensive oceanographic measurements in the ray path region were made simultaneously as part of the Arctic internal wave experiment (AIWEX). The depth-time sampling of the acoustic field satisfied the Nyquist criteria. Preliminary analysis indicates a weak fluctuating environment. The acoustic phase is geometric with variations due mostly to the mean sound-speed profile. The internal wave field is weak, the amplitude and travel time fluctuations are much smaller than those observed in the open ocean, and the statistics are not stationary. These acoustic observations, where the statistics of the medium are well known, will provide a thorough test of available predictions of WPRM in depth, range, and time. [Supported by ONR code 425AR, NORDA, and Marconi Systems Ltd.]

11:35

EE13. Analysis of high-frequency cw tones propagated in the Arctic Ocean. Josko Catipovic and Arthur B. Baggeroer (MIT/WHOI Joint Program in Oceanography, MIT, Cambridge, MA 02139)

During the marginal ice zone experiment (MIZEX), measurements were made of the fluctuating phase and amplitude of a densely spaced set of tones centered around 50 kHz. Eight tones were transmitted simultaneously; the intertone spacing varied from 50 Hz to 1 kHz. The transmitter and receiver were located 10 m below the surface, immediately below the ice. The propagation path was approximately 1.5 km long. The received tones show extensive fading at time rates corresponding to mean drift which produces a nearly constant phase rate; ice motion at approximately 1/10 Hz; and internal waves at higher rates. The tone fading is nearly frequency independent at lower rates, but has a frequency correlation of several hundred Hertz at fade rates of approximately 1 s. The various rates of fluctuation are separated in the analysis to identify the generation mechanism. Envelope and phase amplitude analysis indicate a fully saturated path with most of the effect arising from the higher fluctuation rate. Time-frequency correlation of about 1 s and 300 Hz dominate. For underwater communications these results show that phase coherence at these frequencies is difficult to attain even at close ranges, and that the time-variant channel behavior requires either high coding levels or adaptive code selection or decoding for useful data transmission.

11:50

EE14. A relationship between ocean circulation and volume reverberation in the subarctic northeast Pacific Ocean (Gulf of Alaska). David G. Browning (Naval Underwater Systems Center, New London, CT 06320), R. G. Turner, and J. W. Powell (Defense Research Establishment Pacific, FMO, Victoria, BC, V0S IB0 Canada)

Earlier investigations have shown a significant change in integrated scattering [J. A. Scrimger and R. G. Turner, J. Acoust. Soc. Am. **54**, 483–493 (1973)] and spectral characteristics [R. P. Chapman *et al.*, J. Acoust. Soc. Am. **56**, 1722–1734 (1974)] when transmitting into the subarctic (above 40 north latitude) northeast Pacific Ocean. An analysis of an extensive series of volume reverberation measurements obtained by Turner indicates a strong influence of the counterclockwise circulation around the Alaskan Gyre on the distribution of scattering strengths. At higher frequencies (5–20 kHz) the greater scattering strengths are found in the relatively warm California undercurrent water which flows around the perimeter of the gyre. At lower (1.25–5 kHz) frequencies the greater scattering strengths are found in the relatively cold water such is found in the upwelled subarctic water at the center of the gyre. This implies a significant change in the type of scatterers between these frequency domains. [Work supported by ONT and DREP.]

Joint Meeting of Accredited Standards Committees S3 and S1

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

Meeting of Accredited Standards Committee S3 on Bioacoustics

L. A. Wilber, Chairman S3
422 Skokie Boulevard, Wilmette, Illinois 60091

Standards Committees S3, Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years.
The international activities in ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, for which S1 and S3 serve as the U.S. Technical Advisory Groups, will be discussed.

Meeting of Accredited Standards Committee S1 on Acoustics

E. H. Toothman, Chairman S1
Bethlehem Steel Corporation, Room B-238, Martin Tower, Bethlehem, Pennsylvania 18016

Standards Committee S1, Acoustics. Working group chairpersons will report on their progress in the preparation of standards, methods of measurement and testing, and terminology in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of committee reports is encouraged.

Session FF. Biological Response to Vibration I: Aspects of Vibrotactile Pattern Recognition

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

Chairman's Introduction—1:00

Invited Papers

1:05

FF1. Neural correlates of tactile roughness perception. K. O. Johnson, J. R. Phillips, S. Hsiao, K. H. Fasman, and C. E. Connor (Department of Neuroscience and Biomedical Engineering, The Johns Hopkins University School of Medicine, Baltimore, MD 21205)

Tactile perception of surface roughness was investigated in a combined psychophysical and neurophysiological study using 18 embossed, tetragonal dot patterns with varying dot spacings (1.3 to 6.2 mm) and sizes (0.4- to 1.2-mm diameter). Psychophysical subjects explored these surfaces with the pad of the index finger and reported their estimates of subjective roughness. The results were inverted U-shaped functions of spatial frequency. In neurophysiological experiments, exactly the same surfaces were scanned across the distal pads of macaque monkeys while recording from peripheral mechanoreceptive afferents (SA, QA, and PC afferents). Neural coding hypotheses were tested by comparing various features of the responses with the psychophysical results. Mean rate codes fared poorly as did codes based on "rate fluctuation" (summed absolute values of differences between rate maxima and minima). However, the second central moment of firing rate (variance) for the SA afferents matched the psychophysical data closely, accounting for effects of both dot size and dot spacing. The same measure proved less effective for QA afferents and much less effective for PC afferents. [Work supported by NIH grants NS18787 and GM07057.]

1:35

FF2. The perception of tactile distance: Influences of body site, space, and time. Roger W. Cholewiak (Department of Psychology, Princeton University, Princeton, NJ 08544)

Vibrotactile stimuli are often presented to different body sites. Of the many factors that can affect identification of such patterns are the body site tested and the spatiotemporal influences of nearby stimuli. The data obtained by Kaas, Merzenich, and their colleagues indicate that the degree of "cortical magnification" across different body regions is inversely related to cutaneous receptive field size. Consequently, measures influenced by field size should show similar variation. Geldard and Sherrick have shown such a variation in one tactile illusion, cutaneous saltation. In the series of studies to be discussed, magnitude and matching judgments of patterns consisting of stimuli presented to separated sites, either within tactile matrixes or across separate larger contactors, demonstrate that apparent distance on the skin is inversely related to the interstimulus interval between the two stimuli, directly related to the physical separation between the two, and shows a consistent relationship to cortical magnification. [Work supported by NIH grant NS-04775.]

2:05

FF3. Measures of temporal processing of vibrotactile stimuli. Janet M. Weisenberger (Central Institute for the Deaf, 818 South Euclid, St. Louis, MO 63110)

Knowledge about the ability of the tactile system to process temporally varying stimuli is of interest not only for its additions to understanding the general functioning of the tactile system and its comparability to other sensory systems, but also for the design of tactile aids to alleviate blindness and deafness. In the present paper, previous measures of temporal processing of vibrotactile stimuli are summarized and compared with our recent measures of vibrotactile temporal modulation transfer functions. The modulation transfer function has been found to be a useful framework for describing temporal processing in other sensory modalities. In our studies, sinusoidally amplitude-modulated vibratory carrier signals were presented to the thenar eminence. Results indicated variations in the shape and level of transfer functions with carrier frequency, and considerably better performance for sinusoidal carrier waveforms than for noise carriers. The obtained tactile transfer functions are compared to those for auditory stimuli, and are examined in view of results from studies of vibrotactile temporal masking, to help provide a more complete picture of temporal processing in the tactile system. The implications of these results for the design of tactile aids for sensory impairments are discussed. [Work supported by NSF and NIH.]

2:35

FF4. Vibrotactile pattern recognition. James C. Craig and Paul M. Evans (Department of Psychology, Indiana University, Bloomington, IN 47405)

Several factors influence the identification of complex vibrotactile patterns, e.g., masking, attention, and experience. Two processes that may underlie backward masking are temporal integration and interruption. The types of target patterns used in previous studies have made it difficult to separate these two processes. The results of a study using a specially designed set of patterns that varied in the number of line segments they contained and had no intracharacter redundancy suggest that temporal integration is a major factor in masking. Masking can be reduced by spatially separating two patterns, e.g., presenting one to the index and one to the middle finger. This manipulation improves identification of a single pattern; however, identification of both patterns is difficult. Subjects appear to have difficulty in attending to two patterns presented to two fingers simultaneously. This difficulty is greatly reduced if two fingers on opposite hands are used. It has also been shown that experience improves vibrotactile pattern identification. However, comparisons of blind users of a vibrotactile reading aid with sighted subjects receiving training in the laboratory suggest that major changes in performance result from modest exposure to tactile patterns. Improvement appears to result from increased tactile sensitivity and from learning specific features of patterns.

3:05–3:15

Break

Contributed Papers

3:15

FF5. Electrocutaneous syllable recognition using quasiarticulatory coding of stimulus patterns. Hans Georg Piroth (Institut für Phonetik und Sprachliche Kommunikation der Universität München, München, Federal Republic of Germany)

Assuming that the rate of variation of articulatory movements is slower than that of acoustic signals, it is hypothesized that tactile speech transmission systems should be based on articulatory rather than acoustical parameters of speech. The "System for Electrocutaneous Stimulation SEHR-2" which produces current-controlled bipolar electrical impulses with freely variable intervals was used to code tactile equivalents of CV and VC syllables. In experiment 1, patterns were delivered to the left forearm in such a way that "place of articulation" was transformed quasi-isomorphically to "locus of tactile stimulation." Without previous training, identification tests showed good recognition rates with an average of 65.83% for six vowels coded in a two-dimensional space (upper–lower, distal–proximal) and fairly good identification (38.33%) for six (tactile) consonantal places of articulation coded only in the distal–proximal dimension. In experiment 2, differences in "manner of articulation"were transformed to differences in phenomenal "gestalt" of the tactile patterns created by variation of the interpulse intervals. Recognition of (tactile) manner of articulation was poorer (33.75% for each of four consonants), but clearly above chance level.

3:30

FF6. Frequency resolution via single and multichannel tactile transforms. Terry Hnath, Arthur Boothroyd, and Laurie Hanin (Graduate School, City University of New York, 33 West 42nd Street, New York, NY 10036)

Simple synthetic intonation contours, consisting of 100 ms of constant frequency followed by a 100-ms linear frequency fall, were presented to five normal subjects via two vibrotactile transforms: a single channel (temporal) transform, and a multichannel (spatial) transform. Channel spacing in the multichannel transform was 1 cm. Thresholds for the detection of frequency falls were measured using a three interval, forced-choice, adaptive procedure. With the single-channel transform, mean thresholds on the index finger were 0.3 and 0.4 oct for reference frequencies of 100 and 200 Hz, respectively. On the soft tissues of the forearm, however, the threshold for a reference frequency of 200 Hz was approximately 2 oct. The multichannel transform provided thresholds of one channel on the forearm. In an eight-channel display of voice fundamental, a spatial resolution of one channel represents a frequency resolution of roughly 0.3 oct, which is equivalent to that of a single-channel display on the index finger. Improved resolution can be obtained by increasing the number of channels. [Research funded by NIH grant No. 17764.]

3:45

FF7. An order effect in the discriminability of pulse train sequences. Hans G. Tillmann and Hans Georg Piroth (Institut für Phonetik und Sprachliche Kommunikation der Universität München, München, Federal Republic of Germany)

During investigations of basic mechanisms for tactile speech transmission, an order effect was revealed in the discriminability of electrocutaneous pulse train sequences. Experiment 1 uses a 2IAX-one-step discrimination test to show that tactile sequences of nine pulses with a duration of 24 ms each and an interpulse interval of 15, 35, and 55 ms, respectively, are discriminated better if the sequence with the greater IPI is presented first. In an all-step discrimination test with a constant target (IPI = 35 ms), the extent of the effect could be determined (experiment 2): Of two sequences presented as a pair, the second is judged to have the same IPI as the first if it exceeds IPI of the first by 13.9%. Experiment 3 shows that the effect can be reproduced in the auditory mode with analogously constructed sinusoidal pulse trains. On the other hand, the effect breaks down when pulse train sequences of long and constant durations (2.5 s) are used (experiment 4). As an explanation an analog to the "final lengthening" phenomenon in speech is proposed, as well as an explanation in terms of "time order error."

THURSDAY PM

4:00

FF8. The "system for electrocutaneous stimulation SEHR-2." Hans G. Tillmann and Hans Georg Piroth (Institut für Phonetik und Sprachliche Kommunikation der Universität München, München, Federal Republic of Germany)

SEHR-2 is a 16-channel electrotactile stimulation system producing current-controlled bipolar electrical impulses with freely variable intervals. Its main component is a 16-channel stimulus generation device (expandable to 16 × 16 channels) which produces a periodical series of 16 rectangular pulses with programmable duration and amplitude, as well as a variable pulse repetition rate. Since the impulses are supplied to a capacitive resistance which is depolarized via the skin afterwards, it is ensured that there is no residual dc component. Voltage is adapted to skin resistance so that current intensity is held constant. The device can be connected to a computer by a 16-bit interface mastering the stimulus generation device. A software package has been developed that enables (1) computer-interactive execution of calibration procedures, (2) the editing of stimulus lists for different test procedures, (3) the computer-controlled execution of test sessions, and (4) a preliminary data analysis. The application of the system will be illustrated by the description of typical experiments.

4:15

FF9. Comparison of tactual and auditory discrimination of speech. Rebecca E. Eilers, Özcan Özdamar, D. Kimbrough Oller, Edward Miskiel, and Debra Moroff (Department of Pediatrics, University of Miami, Mailman Center for Child Development, P.O. Box 016820, Miami, FL 33101)

The relationship between tactual and auditory perception of speech was investigated by presenting two speech continua to subjects via a 32-channel computer controlled electrocutaneous display and via normal audition. The stimuli were synthetic /a/ to /ə/ and /sta/ to /sa/ with 9 equally spaced intermediate stimuli between each endpoint yielding two 11-step continua. Five well-practiced adults performed three tasks: (1) an adaptive discrimination task using each of the endpoints as target stimuli; (2) a standard identification procedure (all 11 steps categorized as either endpoint 1 or 11); and (3) a same–different task (equal interval stimuli were discriminated across the continuum). For tactual discrimination, channel information was presented in three spectral configurations conforming to (1) logarithmic, (2) linear, and (3) average (arithmetic mean of log and linear) filtering. Results indicate (a) a close correspondence between auditory and tactual perception, (b) subphonemic discrimination in both modalities, (c) categorical perception in both modalities, and (d) a more auditorylike pattern with log and average filter configurations than with linear. Implications in terms of theories of speech perception will be discussed.

4:30

FF10. Vibrotactile sensitivity of hands occupationally exposed to vibration. J. E. Piercy, A. J. Brammer (Division of Physics, National Research Council of Canada, Ottawa, Canada K1A 0R6), and P. L. Auger (Centre Hopitalier Université Laval, Ste. Foy, Québec, Canada)

The threshold of vibrotactile perception at the finger tip has been measured for six forest workers who regularly use a chain saw, and, for comparison, ten laboratory workers with no significant vibration exposure. All subjects were carefully screened for confounding medical variables, such as carpal tunnel syndrome and diabetes. The measurements were performed over the frequency range of 4–400 Hz in the laboratory to permit careful control of physical variables, such as orientation of the vibrating probe with respect to the finger tip, contact force, and hand movement. Preliminary analysis of our results, together with those of others, suggests that the neurological component of the hand-arm vibration syndrome develops first as a change in the threshold of Pacinian corpuscles, which later spreads to the other cutaneous mechanoreceptors.

4:45

FF11. Auditory and oral tactile sensory system interactions: Magnitude estimation and cross-modality matching. Donald Fucci (Ohio University, Athens, OH 45701), Linda Petrosino (University of North Carolina, Chapel Hill, NC 27514), and Daniel Harris (Brown Schools, Ranch Treatment Center, Austin, TX 78745)

Research that has been done on the interaction of auditory and oral tactile sensory systems has involved experimental techniques intended to disrupt one or both sensory modalities. An instrumentation system has been designed to investigate the relationship between auditory and oral tactile sensory processes without disruption of one or both sensory channels. This instrumentation can be employed to obtain judgments of auditory and oral tactile sensation magnitudes by the psychophysical scaling methods of magnitude estimation and cross-modality matching. Magnitude estimation scaling requires the subject to assign numbers to subjectively match the intensities of different stimulus levels presented by the experimenter. Cross-modality matching requires the subject to adjust the stimulus intensities applied to one sensory modality to match those applied to another sensory modality by the experimenter. The instrumentation system was tested through a pilot investigation in which lingual vibrotactile and auditory scaling behavior was determined for a group of 20 normal speakers and a group of 10 stutterers. Results suggested that the instrumentation, utilizing the psychophysical scaling methods of magnitude estimation and cross-modality matching, may serve as a useful research tool for investigating the possible iterations of auditory and oral vibrotactile sensory processes.

THURSDAY AFTERNOON, 15 MAY 1986 — RITZ ROOM, 1:30 TO 4:30 P.M.

Session GG. Engineering Acoustics III: Ultrasonic Sensing in Air—Demonstrations

Lee N. Bolen, Chairman
Physical Acoustics Research Group, Department of Physics and Astronomy, University of Mississippi, University, Mississippi 38677

Dale Bazill, Vice Chairman
Projects Unlimited, Inc., 3680 Wyse Road, Dayton, Ohio 45414

Demonstration Session

The speakers from Session AA and other workers in the field will demonstrate systems and devices for ultrasonic sensing in air.

Session HH. Musical Acoustics II

Walter E. Worman, Chairman
Department of Physics, Moorhead State University, Moorhead, Minnesota 56560

Chairman's Introduction—2:00

Contributed Papers

2:05

HH1. A music synthesis system using a personal computer, drum sensor, and MIDI-controlled synthesizer. D. Barr, J. Kohut, and M. V. Mathews (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A music synthesis system has been built. It consists of an AT&T 6300 computer connected to a Yamaha DX-7 synthesizer and to an electronic drum sensor on which the performer plays. With the exception of the electronic drum, all components in the system can be purchased, the total cost being about $5000. Device driver and interrupt handler programs have been written to interface the computer with the synthesizer and sensor hardware. A real-time control program, RTLET, has been written. In RTLET a set of real-time control processes are activated by sending dated letters between processes. A postmaster program, responsible for overall timing, receives and delivers letters to a control process and activates the process at the date time specified on the letter. RTLET has been used to implement conductor control processes in which the performer controls the playing of a score stored in the computer memory by means of drum strokes.

2:20

HH2. Handlebars for the Yamahas: The struggle toward more intuitive programming of FM synthesizers. G. L. Gibian, E. N. Harnden (Physics Department, American University, Washington, DC 20016), and D. L. Bort (Johns Hopkins Applied Physics Laboratory, Johns Hopkins Road, Laurel, MD 20707)

One currently popular technique for synthesis of musical sounds employs audio-rate frequency modulation (FM). Although this technique was first suggested in the 1970s [Chowning, J. Audio Eng. Soc. **21**, 526–534 (1973)], it is still somewhat difficult to control. The specification of a particular tone color in terms of traditional Fourier amplitudes remains problematic, especially for time-varying spectra because the relative amplitudes of all the components are governed by Bessel functions. Conversely, the natures of the spectra resulting from various configurations of carriers, modulators, and modulators of modulators are not intuitively obvious. Preliminary work on a Fourier-to-Bessel mapping program which is intended to simplify FM implementations of specific spectra will be reported.

2:35

HH3. Methods of articulation in piano performance. Caroline Palmer (Psychology Department, Cornell University, Ithaca, NY 14853)

Pianists of varying levels of experience performed polyphonic excerpts from the classical repertoire on a computer-monitored (MIDI interface) synthesizer. Preliminary findings showed consistent and reliable temporal patterns of articulation. Both legato and staccato patterns were accurately predicted by a combination of the durations of successive notes in the musical score. Onset times of the individual voices notated as simultaneous differed such that the melody preceded the other voices, similar to Rasch's [J. Acoust. Soc. Am. **43**, 121–131 (1979)] finding with ensemble players. Rubato patterns (alterations in tempo from mechanical regularity) were also consistent within an excerpt, demonstrating predictable changes within phrases and at cadences. Patterns of rubato in these polyphonic performances were similar to Bengtsson and Gabrielsson's [Proc. R. Swed. Acad. Music **39**, 27–60 (1983)] profiles of monophonic performances by pianists. Each of these patterns of articulation was strongest in experienced pianists, and existed to varying degrees in student musicians. When experts were asked to perform nonmusically, the articulation patterns were still present but dampened in degree. Ongoing research is comparing the influence of amplitude (using a velocity-sensitive weighted keyboard) on pianists' articulation patterns. [Work supported by NSF and NIMH.]

2:50

HH4. Timbre discrimination of musical instruments in a concert hall. Pamela J. Goad and Douglas H. Keefe (Systematic Musicology, School of Music, DN-10, University of Washington, Seattle, WA 98195)

Isolated tones and melodies of six tones were recorded in different locations of a concert hall using five musical instruments played in their upper and lower registers. Each trial consisted of either a pair of tones or a pair of melodies recorded simultaneously at different locations on one instrument. The room transmission path to each location varied. Using a monaural headphone presentation, the task was to determine whether the pair was the "same" or "different" with respect to timbre. To prevent subjects from discriminating based upon room variation in background noise, it was necessary to add pink noise below 2700 Hz; however, the signal-to-noise ratio remained above 30 dB at all frequencies. Our results showed that subjects could discriminate timbre in this context. No significant differences were found between isolated tones and melodies, or between registers. Significant differences appeared between instruments with subjects best able to discriminate flute timbre followed in rank order by clarinet, trumpet, viola, and marimba. Work is in progress to compare results with theories of timbre perception.

3:05

HH5. The consonance of Pierce scale dyads and triads. M. Chowning, L. A. Roberts, and M. V. Mathews (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A study of the judged consonance of dyads and triads formed from chromatic intervals in the Pierce scale was carried out. One group of subjects listened to dyads in which the maximum interval was 16 steps or 3 steps more than a tritave (13 steps) and the minimum interval was 1 step. Results showed that: (1) the tritave is very consonant; (2) a single-step interval is very dissonant; (3) intervals of 4 and 6 steps, which make up the "major" and "minor" Pierce scale chords, are relatively consonant; and (4) other intervals have a pattern of consonance which is not yet fully understood. Results also showed that the judged consonance is independent of the average frequency of the dyad. Another group of subjects judged the consonance of Pierce scale triads as well as the dyads which comprised these triads. An additive model gave an excellent fit to the data, that is, the perceived consonance of individual dyads contained in a triad predicts judgments for the triad.

3:20

HH6. Mapping musical intervals to affective qualities: A projective study. Scott Makeig (Children's Hospital Research Center, 8001 Frost Street, San Diego, CA 92123)

Many claim that the various musical intervals used in Western music create identifiable affective impressions which musicians use for musical communication. Alain Danielou has proposed, furthermore, that these affective impressions form a space with a three-dimensional structure isomorphic to the natural three-dimensional structure of the intervals themselves (ratios composed of positive and negative powers of 2, 3, and 5). To test this hypothesis, 36 Marathi-speaking college students at Ahmednagar College, Maharastra, India, were asked to imagine that each of 24 complex-tone melodic intervals was the "sound-name" of a person in some distant village. They were then given 36 bipolar adjective pairs and asked to rate the personality of the person they imagined on five-point scales. (Previous studies have shown that subjects make affective judgments about sounds far more easily when imagining personalities than when rating the sounds themselves.) Results indicate that Indian students associate the main intervals used in Western music with separable personalities, and that these personalities compare well with those Western listeners reported in a previous study. [Work supported by the American Institute of Indian Studies.]

THURSDAY AFTERNOON, 15 MAY 1986 — SAVOY ROOM, 1:30 TO 5:00 P.M.

Session II. Physical Acoustics VII: Scattering: Liquids and Solids

Dale Chimenti, Chairman

NDE Group, Air Force Materials Research Laboratory, AFWAL/IMLLP, Wright Patterson Air Force Base, Ohio 45433

Contributed Papers

1:30

II1. Acoustic scattering in an ocean environment: II. Shallow water applications. Gary S. Sammelmann and Roger H. Hackman (Naval Coastal Systems Center, Panama City, FL 32407)

At a previous meeting of the Acoustical Society of America [R. H. Hackman and G. S. Sammelmann, J. Acoust. Soc. Am. Suppl. 1 78, S76(1985)] we presented a transition matrix formalism for the acoustic scattering in an inhomogeneous waveguide. Here, we report on an application of this formalism to the acoustic scattering from elastic targets in a range-independent shallow water environment. Particular emphasis is placed on identifying features in the backscattered wave which are unique to this environment and on elucidating the underlying physical mechanisms responsible for these features. [Work supported by the Office of Naval Research.]

1:45

II2. Scattering from submerged objects near an interface. M. F. Werby and R. B. Evans (NORDA, Code 221, NSTL, MS 39529)

Scattering from bounded objects in a free ocean has been treated extensively in recent years with considerable progress being made to date. Severe complications, however, arise for objects either in a waveguide or in the presence of an interface. It is possible to formulate this problem exactly and we present results for the case of an object near a sound-soft or sound-hard interface. Results are presented for objects as they vary in distance from an interface. Angular distributions are particularly suited for such a study where results are compared with the free ocean case. A comparison of the two cases shows that an object near an interface includes effects not seen in the free ocean example, namely reflections of the refracted wave from the interface and reflections from the object into the bottom and back toward the source of the field. This leads to angular distributions that typically have four peaks as opposed to two peaks from the free ocean case. The exceptions to this case are when the field is either parallel or perpendicular to the interface. The first of these exceptions (as expected) shows little difference with the free ocean case while the latter case is dominated by reflections from the interface itself yielding a large backscattered return.

2:00

II3. On conversion of Lamb waves to leaky Lamb waves in a plate partially immersed into the liquid. S. I. Rokhlin (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

The problem of Lamb waves diffraction on the boundary of separation of the free and immersed (into the liquid) parts of the elastic plate is theoretically analyzed. In the theoretical model, it is assumed that the free part of the plate is separated from the liquid by infinitely thin air barriers. It is shown that in the first approximation the reflection coefficient is proportional to the attenuation factor of the leaky Lamb waves. The diffracted field into the liquid is formed by a cylindrical wave which is generated on the boundary line and by leakage of the energy carried by leaky Lamb waves in the plate. Analysis of the leakage field shows an anomalous behavior of the backward Lamb wave. This backward wave has negative dispersion and therefore the angle of the energy leakage for this wave (angle in Snell's law) must be taken as negative. It is shown that for calculation of the radiation field, the Kirchoff approximation can be used.

2:15

II4. Application of the integral equation method to the calculation of acoustic scattering by a space launcher. P. Malbéqui, S. M. Candel,[a] and E. Rignot[b] (Office National d'Etudes et de Recherches Aérospatiales, B. P. 72, 92322 Châtillon Cedex, France)

The objective of this paper is to illustrate the possibilities of the integral equation method in the analysis of acoustic scattering by the structure of a space launcher. Such an analysis is important in situations where the sound field reaches critical levels at take-off. In these circumstances an increase in amplitude associated with reflections and scatterings must be estimated with precision. The method is based on integral representations of the wave field and only requires a surface mesh of the solid structure. The number of nodes of such surface meshes is much smaller than the 3-D finite difference or finite element equivalents. Another important advantage of the method is that it automatically incorporates the farfield radiation condition. This condition is only approximated in field methods. The calculations are performed with a code originally developed by Hamdi [M. A. Hamdi, Thèse de doctorat d' Etat, Université de Technologie de Compiègne (1982)] and validated at ONERA for the scattering of a plane wave by a hard sphere. Our presentation concerns more specifically the Ariane 4 fairings (the payload housing). It is found that the geometrical shape of the fairings causes local increases of the sound field. This tendency is confirmed by measurements performed on a 1/20 model of the launcher. [Research supported by Centre National d'Etudes Spatiales.] [a] Also at Ecole Centrale des Arts et Manufactures, Chatenay-Malabry, France, [b] Student of Ecole Centrale des Arts et Manufactures.

2:30

II5. Ray trace calculations of ultrasonic fields. J. A. Johnson, N. M. Carlson, and D. M. Tow (Idaho National Engineering Laboratory, P. O. Box 1625, Idaho Falls, ID 83415)

A computer code has been written to calculate realistic ultrasonic fields in solids. The program calculates the field due to a transducer coupled to the solid through either a fluid medium such as water or a solid medium such as a plastic shoe with a thin layer of fluid couplant. In this technique the transducer face is divided up into many small areas, each of which is assumed to be a source of spherical waves. A ray is traced from each source point, through the coupling medium and into the solid to the requested field point. The vector sum of the fields of each of the rays is then calculated to find the total incident field at that point for a particular frequency. The calculations have been compared to the fields predicted by the models of Thompson and Gray [R. B. Thompson and T. A. Gray, "Analytic Diffraction Corrections to Ultrasonic Scattering Measurements," in *Review of Progress in QNDE 2,* edited by D. O. Thompson and D. E. Chimenti (Plenum, New York, 1983), pp. 567–586] and Thompson and Lopes [R. B. Thompson and E. F. Lopes, "The Effects of Focusing and Refraction on Gaussian Ultrasonic Beams," J. Nondestruct. Eval. **4**, 107–123 (1984)] and with experimental results. [Work supported by the U. S. Department of Energy, Office of Energy Research, Office of Basic Energy Sciences, under DOE Contract No. DE-AC07-76ID01570.]

2:45

II6. Acoustic radiation from an impulsive point source in an anisotropic solid. J. H. M. T. van der Hijden (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108)

The wave propagation problem of calculating the radiation of elastic waves from a source in an anisotropic medium was formally solved by Duff [G. F. D. Duff, Philos. Trans. R. Soc. London Ser. A **252**, 249–273 (1960)], by using the standard method of multiple Fourier transforms. Duff's solution however contains integrals over the wave surface and integrals over the volume between the wave surface and the convex envelope of that surface. Because these wave surfaces can have cusps, Duff's solution is unsuitable for numerical calculation. The farfield approximate solution is known explicitly [E. A. Kraut, Rev. Geophys. **1**, 401–448 (1963); and M. J. Lighthill, Philos. Trans. R. Soc. London Ser. A **252**, 397–470 (1960)]. It uses the Gaussian curvature of the slowness surface. Even this approximate solution is quite involved when computed numerically. We present the solution as derived by using the Cagniard–de Hoop method. The complexity of the solution is now reflected in the Cagniard–de Hoop contours that represent the mapping from real time to complex ray parameter. These contours can be computed quite easily. Therefore the final expressions are not only very elegant but also computationally friendly. We will show some examples of the results.

3:00

II7. Theoretical analysis of the angle beam ultrasonic testing method of unidirectional graphite/epoxy composite material. S. I. Rokhlin, T. K. Bolland, and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

The reflection and refraction of an ultrasonic beam on the boundary of separation between a coupling medium and a unidirectional composite material is studied. The composite material is considered as an anisotropic material of hexogonal symmetry. Two variants of the coupling are studied: first is the immersion variant, when the ultrasonic beam is incident from the liquid; second is the contact variant, when the ultrasonic beam is incident from an isotropic solid wedge and the slip boundary condition, between the solid wedge and the composite material, is used. The energy reflection and refraction coefficients are analyzed as a function of the angle of incidence, with the angle of the orientation of the fibers relative to the incidence plane as a parameter. Principal attention is given to critical-angle behavior and the deviation of the energy flux from the wave normal in the composite material.

3:15

II8. Guided waves in nonuniform media. Ian M. Lindevald and A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

A waveguide is characterized by three parameters. These are the waveguide cross section $A(z)$, the medium's inertia (density) $\mu(z)$, and its elasticity $\sigma(z)$. The coordinate system is defined by the positions of the guided wave's (constant z) isophase surfaces. The one-dimensional waveguide equations for generalized effort ϵ and flow ϕ can be written $\ddot{\xi} = (\sigma/\mu)\,[\xi'' + 2(\ln R_\xi)'\xi']$, where ξ is either ϵ or ϕ, a dot denotes $\partial/\partial t$ and a prime $\partial/\partial z$. In the approximation that the wavefronts are parallel, $R_\epsilon = (A/\mu)^{1/2}$ and $R_\phi = (\sigma/A)^{1/2}$. For non parallel wavefronts greater accuracy can be gained by redefining the R's in terms of the M and N functions of Weibel's waveguide equations. The equation for each aspect (ϵ,ϕ) leads to its own locally defined phase and group velocities $v_{p\xi}$ and $v_{g\xi}$. However, the local wave impedance Z_ω and the rate of energy transport v_ω are defined via both aspects jointly. In general, $v_{p\xi}, v_{g\xi}, v_\omega$, and Z_ω depend on all three parameter functions. If all three functions are constant, we find the familiar results $Z_\omega = (\sigma\mu)^{1/2}/A$ and $v_{p\xi} = v_{g\xi} = v_\omega = (\sigma/\mu)^{1/2}$. When $A(z)$ varies, leaving $\mu(z)$ and $\sigma(z)$ constant, the equation reduces to the familiar "Webster" equation (or modified by Weibel) first studied by D'Alembert, Bernoulli, and Euler. Otherwise the wave parameters are expressible in terms of the waveguide's generalized radius R_ξ, its taper R'_ξ, and its flare R''_ξ. [Work supported by NSF and NIH.]

3:30

II9. Transition from sound-soft to rigid behavior in scattering from submerged elastics shells over a broad frequency range. M. F. Werby (NORDA, Code 221, NSTL, MS 39529) and G. Gaunaurd (Naval Surface Weapons Center, White Oaks, Silver Spring, MD 20910)

We have determined previously that at suitably high frequencies elastic shells for a variety of thicknesses behave rigidly in the absence of narrow resonances. This observation was noted for thicknesses ranging from 1% to 2.5% for aluminum, steel, and WC for frequencies out to *ka* values of 300. Moreover, we have presented results at the low-frequency end (low *ka*) that showed a distinct sound soft behavior for very thin shells in the absence of a resonance. This behavior persisted out to moderate values of *ka*. The procedure to determine whether a particular background was manifest to twofold. Firstly, the rigid or sound-soft response is subtracted in appropriate partial wave space from the elastic response to determine the residual response. The residual response shows mainly resonance contributions at the resonance frequencies with little contribution otherwise. One can also examine the relative phase between the expected background and the elastic response and it should vary slowly except at a response where it varies rapidly with changing phase (a change of at least 180 deg). We here choose the case of a steel shell of thickness 0.3% of the radius of the shell and present results for the *ka* range from 0.0 to 500. Here the background is demonstrated to vary from sound-soft at the lower end of *ka* to rigid at the upper end with no clear background inbetween.

3:45

II10. The eigenfrequencies of spheroids obtained from an integral phase matching condition. Barbara L. Merchant, Y. J. Stoyanov, A. Nagl, H. Überall (Department of Physics, Catholic University, Washington, DC 20064), S. H. Brown, and J. W. Dickey (David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402)

The complex acoustic eigenfrequencies of impenetrable spheroids have been found previously by satisfying the boundary conditions with spheroidal wavefunctions [J. M. D'Archangelo *et al.*, J. Acoust. Soc. Am. 77, 6 (1985)]; alternately, they are obtained here by using a T-matrix code. However, a physical explanation of the eigenfrequencies can be given in terms of surface waves circumnavigating the spheroid along a geodesic, and a phase matching condition can be written down which takes the form of an integral, due to the variations of the surface wave propagation constant along the path. A model is used in which the propagation constant is approximated by the known one on a sphere, with the same local radius of curvature along the propagation direction. This condition serves either to obtain the complex eigenfrequencies, or to test the accuracy of eigenfrequencies obtained by other methods. [Supported by NSRDC and ONR.]

4:00

II11. Space-time duality for scattered wave inversion by backpropagation. Robert P. Porter (Electrical Engineering Department FT-10, University of Washington, Seattle, WA 98195)

In ultrasonic and acoustic diffraction tomography, objects are scanned over a wide range of illumination angles in contrast with seismic inversion where coverage is limited but wideband data are available. In this presentation it is shown that either wide angle or wide bandwidth illumination is sufficient for determining the structure of a weak scatterer when the scattered field is backpropagated or migrated toward the object. The backpropagation concept is briefly reviewed and generalized to arbitrary recording geometries. After finding the spatial spectra of the backpropagated field, we shall see that the object scattering function can be extracted by scanning an illuminating plane wave over a wide frequency range or over all possible directions. Two recent inversion methods, holographic tomography (monochromatic waves and wide angle coverage) and image filtering (single wide band plane wave) will be discussed as examples of backpropagating methods that exploit space–time duality.

4:15

II12. Bistatic cross sections of thin spherical elastic shells at resonance, and the resonance order. Michael F. Werby (Naval Ocean Research & Development Activity, NSTL, MS 39529) and Herbert Überall (Physics Department, Catholic University, Washington, DC 20064)

The acoustic scattering amplitude of elastic objects separates into specular-reflection terms, and elastic-resonance terms [L. Flax *et al.*, J. Acoust. Soc. Am. 63, 723 (1978)]. For separable geometry, this holds for the individual mode amplitudes separately, and the problem arises how to classify the oberved resonances as to their mode order. For cylindrical scatterers at normal incidence, where the modal amplitudes contain $\cos n\phi$, the experimental solution is provided by observing the corresponding "daisy" pattern of the resonant bistatic cross section after isolating it by the MIIR method [G. Maze and J. Ripoche, J. Acoust. Soc. Am. 73, 41 (1983)]. We demonstrate here, for the case of spherical aluminum shells of thickness 0.1%, that mathematically, the order of a resonating mode is determined by coherent subtraction of the soft-sphere nonresonant background, and observation of the remaining bistatic resonant cross section whose pattern is determined by P_n $(\cos \theta)$. [H. Überall was supported by the Office of Naval Research.]

4:30

II13. Synthesis of backscattering from an elastic sphere using the Sommerfield–Watson transformation and giving a Fabry–Perot analysis of resonances. Kevin L. Williams (Naval Coastal Systems Center, Code 4120, Panama City, Florida 32407) and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

The Sommerfield–Watson transformation was recently applied to the acoustical backscattering from elastic spheres in water having $ka \gg 1$. Expressions for the scattering due to each class of elastic surface wave (e.g., the Rayleigh wave) were interpreted in terms of contributions from repeated circumnavigations. In this presentation these expressions are summed in closed form as in the analysis of Fabry–Perot resonators. This closed form exhibits resonance behavior as is shown for the particular case of the Rayleigh wave on a tungsten carbide sphere. The sphere's form function is synthesized by adding the Fabry–Perot type contributions to the specular reflection. The procedure is confirmed by comparison with the exact backscattering form function magnitude $|f|$ for an aluminum sphere is the range $10<ka<30$ and a tungsten carbide sphere $10<ka<80$. It is also shown that in the tungsten carbide case the interference of the specular and Rayleigh contributions produce the underlying structure in $|f|$ while the whispering gallery wave resonances produce a finer superposed structure. Phase shifts are identified which affect the signature in $|f|$ of the Rayleigh-wave resonances. [Work supported by ONR.]

4:45

II14. Iterative inverse scattering method employing Gram–Schmidt orthogonalization. W. Tobocman (Physics Department, Case Western Reserve University, University Circle, Cleveland, OH 44106)

An inverse scattering method is tested which appears to work very well. It is an iterative procedure based on the distorted wave Born approximation. The resulting Fredholm equation of the first kind is solved by a projection method that requires the construction of an orthonormal set of basis functions from the set of kernel functions. This can be done by the spectral expansion method which entails matrix diagonalization or by Gram–Schmidt orthogonalization. The method is tested on a simple one-dimensional optical wave inverse scattering problem. Both orthonormalization methods lead to a rapidly convergent and stable iteration. The spectral expansion method gives somewhat better results, but the Gram–Schmidt orthogonalization method, which is the novel aspect of our approach, is much less time consuming. We show how the projection method can be generalized to the case of acoustic wave scattering where the target must be characterized by more than a single independent function of position.

Session JJ. Physiological Acoustics VI and Psychological Acoustics VII: Frequency Discrimination and Coding

Frederic L. Wightman, Chairman
University of Wisconsin, Waisman Center, 1500 Highland Avenue, Madison, Wisconsin 53706

Contributed Papers

1:15

JJ1. Frequency and practice interactions in frequency discrimination. Lynne Werner Olsho and Elizabeth G. Koch (Department of Otolaryngology, Box 430, University of Virginia, Charlottesville, VA 22908)

Differences between human infants and adults in frequency discrimination resemble differences between untrained and trained adults: frequency difference limens (FDL's) tend to be relatively worse for infants and untrained adults at low frequencies [L. W. Olsho, Inf. Behav. Dev. 7, 27–35 (1984)]. To demonstrate directly that the effect of training in frequency discrimination depends on frequency, the course of improvement in frequency discrimination was examined in adults at 500, 1000, and 4000 Hz. Stimuli were pure tones presented at a 40-dB sensation level. A two-alternative forced-choice procedure was used, and FDL was tracked using a one-up, two-down adaptive procedure. Four subjects completed 200–250 trials of training at each frequency. For these subjects not only was initial FDL much higher relative to final FDL at the two lower frequencies, but tracking width decreased progressively with increasing frequency. Four additional subjects were tested for a total of 20 h each to make certain that asymptotic performance was reached. The final FDL's obtained by these listeners were similar to those reported by other investigators [e.g., C. C. Weir *et al.*, J. Acoust. Soc. Am. **61**, 178–184 (1977)], and the same practice effects were observed as in the first group. [Work supported by NIH.]

1:30

JJ2. Detecting spectral and temporal target differences in the harmonic complex. Jane E. Daniel and Michael Kubovy (Rutgers University, Department of Psychology, Tillett Hall, Kilmer Campus, New Brunswick, NJ 08903)

We created pairs of harmonic complexes identical except for one harmonic (the target). Varying target amplitude and/or phase resulted in stimulus pairs which were different either in the frequency domain (spectral target difference) or in the time domain (temporal target difference). A signal detection analysis was applied to subjects' detection of spectral and temporal target differences at low through high harmonics. Spectral target differences were detectable at low harmonics only. Temporal target differences were detectable over a wider range of harmonics than previously claimed [H. Duifhuis, J. Acoust. Soc. Am. **48**, 888–893 (1970)]. Using an adaptive psychophysical method we found that the number of low harmonics where spectral target differences were detectable varied with fundamental frequency. This result is not consistent with Duifhuis' (1970) observations, but does follow quantitative predictions based on the critical bandwidth, as well as other findings [R. Plomp, *Aspects of Tone Sensation* (Academic, London, 1976)].

1:45

JJ3. Relationship between frequency selectivity and frequency discrimination measured in subjects with unilateral and bilateral cochlear impairments. Brian C. J. Moore and Brian R. Glasberg (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

These experiments were designed to test a model proposed by Zwicker [E. Zwicker, Acustica **6**, 356–381 (1956)] in which the frequency difference limen (DLF) for a pure tone is assumed to be determined by the slope (S) of the excitation pattern at the point where it is steepest (on the low-frequency side), and by the size of the smallest detectable change in excitation at that point (DLI): DLF = DLI/S. S was estimated by measuring the auditory filter shape using a notched-noise maker. The DLF was measured in two tasks: pulsed tones frequency discrimination, and detection of frequency modulation. The DLI was measured in two comparable ways, but a high-pass noise was added to the tone to mask the upper side of the excitation pattern. Subjects had moderate unilateral or bilateral cochlear impairments. Tones were presented at 80 dB SPL. For the modulation-detection task the values of the DLF's predicted by the model were comparable to those actually obtained, although the predicted and obtained values were not highly correlated. For the pulsed-tones task the obtained DLF's were consistently smaller than predicted by the model. Overall the data suggest that Zwicker's model cannot account *both* for the detection of frequency modulation and for the frequency discrimination of pulsed tones. [Work supported by the Medical Research Council, U.K.]

2:00

JJ4. Discrimination of frequency ratios. Deborah A. Fantini and Neal F. Viemeister (Department of Psychology, University of Minnesota, Minneapolis, MN 55455)

Recent work on spectral profile analysis demonstrates that observers can extract amplitude relationships between components in complex stimuli. The experiments to be discussed demonstrate that observers also can extract frequency relationships, specifically, they can discriminate frequency ratios of components in a two-tone complex. The observer is presented with two complexes, one whose components are at frequencies of f and fr, the other with components at f and $f(r + \Delta r)$, and, in a 2IFC task, is required to choose the observation interval containing the larger ratio. The base frequency f is random both within and across trials; thus, the observer must, in effect, compare the frequencies within each complex to perform well. Good performance is shown with small values of Δr: for example, with $r = 1.25$ a Δr of 0.017 yields approximately 71% correct when f is randomized over a 400–800 Hz range. This is comparable to the performance shown by trained musicians in discrimination of musical intervals when each of the two tones comprising the interval are presented successively [E. M. Burns and W. D. Ward, J. Acoust. Soc. Am. **63**, 456–468 (1978)]. The paper will summarize our initial findings on the effects of varying r and frequency range, and on the form of the psychometric function. [Supported by NINCDS grants NS12125 and NS07889.]

2:15

JJ5. Frequency-based and temporal-based estimates of auditory filter bandwidth. Therese M. Velde and Joseph W. Hall (Auditory Research Laboratory (Audiology), Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

Estimates of auditory filter bandwidths were examined from measurements of frequency selectivity and temporal acuity. Frequency selectivity was measured using psychophysical tuning curves and comb-filtered noise masking patterns. Temporal acuity was measured using gap detection and temporal masking period patterns. Measurements for all four experimental paradigms were made at 500 and 3500 Hz using both normal-hearing and hearing-impaired listeners. The results are discussed in

terms of the relation between frequency selectivity and temporal acuity as predicted by the auditory linear filter model. The results of the hearing-impaired listeners are compared to the results of the normal-hearing group and are discussed *re*: the predicted relation between frequency selectivity and temporal acuity.

2:30

JJ6. Effect of forward masker duration on measures of frequency selectivity. Sid P. Bacon and Walt Jesteadt (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

The effect of forward masker duration on measures of frequency selectivity was examined in two experiments. Masker duration was 50 or 400 ms, and signal duration was 20 ms, with no delay between masker offset and signal onset. In the first experiment, growth-of-masking functions were measured for four normal-hearing subjects for masker frequencies below, at, or above the 1-kHz signal frequency, and straight lines were fitted to those functions using a least-squares criterion. Psychophysical tuning curves (PTC's) and input filter patterns (IFP's) were derived from the fits. In the second experiment, masking patterns (MP's) were measured for a 1-kHz masker presented at 50, 70, and 90 dB SPL. All three measures (PTC's, IFP's, and MP's) show better frequency selectivity for the 400-ms masker, particularly on the high-frequency side for the PTC's and IFP's, and on the low-frequency side for the MP's. These data are consistent with a sharpening of frequency selectivity at a stage prior to the adaptation observed in forward masking. [Work supported by NIH.]

2:45

JJ7. Mapping auditory phase sensitivity with timbre discriminations. Roy D. Patterson (MRC Applied Psychology Unit, 15 Chaucer Road, Cambridge CB2 2EF, England)

A flat-spectrum periodic sound composed of 31 equal-amplitude harmonics was used to investigate the effects of phase on timbre. Listeners discriminated a pulse train, wherein the harmonics start in cosine phase from (1) nonmonotonic alternating-phase waves wherein successive harmonics have fixed phases p and q, and (2) monotonic, decelerating-phase waves wherein successive harmonics are phase shifted by an ever decreasing amount. Both phase manipulations produce detectable timbre changes for pulse rates below 400 Hz but monotonic phase increments have to be much larger than alternating phase differences for equivalent performance. For periodic sounds, the output of an auditory filter with center frequency above the fourth harmonic is a modulated sine wave and the modulator's characteristics appear to determine performance. Alternating-phase waves produce a local maximum in the modulator trough which enables the discrimination to be made within each frequency channel. (These waves sound an octave above the pulse train.) Monotonic-phase waves produce strictly sinusoidal modulators and so the discrimination requires detecting modulator phase differences across filters which is more difficult. (These waves sound like damaged pulse trains.)

3:00–3:15

Break

3:15

JJ8. Conditioned pitch change and pitch adaptation. Robert W. Peters (Division of Speech and Hearing Sciences, Department of Medical Allied Health Professions, School of Medicine, The University of North Carolina, Chapel Hill, NC 27514) and Joseph W. Hall (Department of Communicative Disorders, Northwestern University, 303 East Chicago Avenue, Chicago, IL 60611)

In previous studies we have reported long-lasting changes in the pitch of a complex sound following its association with a sound of a different pitch. The present study concerned whether a sound altered in pitch would show pitch adaptation effects commensurate with its pitch or frequency status. The test stimulus was the sixth, seventh, eighth, and ninth harmonics of F_0 196 Hz. The conditioning tone was the first five harmonics of F_0 204 Hz and the adapting stimulus was the first five harmonics of F_0 200 Hz. Stimuli F_0 196 and 204 Hz were heard in association for 30 min each day for 6 days. This resulted in a shift of F_0 196 Hz to a pitch near 204 Hz. The results indicated a pre-conditioning adaptation of the pitch of F_0 196 Hz downward and a post-conditioning upward shift. While the pitch change was in process and near 200 Hz, no adaptation occurred. The results will be discussed with reference to plasticity of the auditory system.

3:30

JJ9. Pitch-intensity effects and model calculation of the level dependence of VIIIth nerve response functions. Arnold Tubis, Kenneth Jones (Department of Physics, Purdue University, West Lafayette, IN 47907), and Edward M. Burns (Department of Speech and Hearing Science, University of Washington, Seattle, WA 98195)

It has been suggested that pitch intensity effects may be correlated with level-dependent shifts in the location of peaks in the VIIIth nerve interspike interval (ISI) histogram [K. Jones, A. Tubis, and E. M. Burns, J. Acoust. Soc. Am. Suppl. 1 **74**, S8 (1983)]. In order to further explore this possibility, we have carried out a detailed numerical simulation of these shifts and compared them with average human pitch-intensity data. We use the conventional stochastic neural firing model in which the conditional probability for a firing in the brief time interval $(t, t+\Delta t)$ subsequent to the last previous firing at time t' is $s(t)r(t-t')\Delta t$, where $s(t)$ is the signal-excitation function and $r(t-t')$ is the neural refractory function. We use the empirical PST response function of D. H. Johnson [Ph.D. thesis, MIT (1974)], the neural refractory function of R. P. Gaumond [Ph.D. thesis, Washington University (1980)], and the technique of K. Jones, A. Tubis, and E. M. Burns [J. Acoust. Soc. Am. **78**, 90–94 (1985)] for extracting $s(t)$ from the PST response. [Work supported by NIH.]

3:45

JJ10. Frequency regionalization in the fish ear. Mardi Cox, Peter H. Rogers (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332), Arthur N. Popper, William M. Saidel (Georgetown University, School of Medicine–School of Dentistry, Department of Anatomy, Washington, DC 20007), and Richard R. Fay (Parmly Hearing Institute, Loyola University of Chicago, Chicago, IL 60626)

The exact mechanism for frequency discrimination by bony fishes is unknown; however, the results of an experimental study by P. S. Enger [*Hearing and Sound Communication in Fishes* (Springer, New York, 1981), pp. 243–255] suggest the existence of frequency regionalization on the saccular macula in the ears of codfish. Frequency regionalization is similar to the place mechanism in the cochlea in that different frequencies stimulate different areas of the saccular and lagenar maculae. In this investigation, goldfish are subjected to a single-frequency tone, about 140–150 dB above threshold, for two hours in order to damage areas sensitive to that frequency. During this exposure, the fish is constrained inside a waveguide with controllable acoustic pressure and particle velocity characteristics. The extent of regionalization on the maculae is determined based on electrophysiologically measured degradation of frequency tuning, and hair cell damage found in examination under a scanning electron microscope. In addition, the separate effects of acoustic pressure and particle velocity on frequency regionalization are compared. [Work supported in part by ONR and NIH.]

4:00

JJ11. Interaction between spectral and periodicity pitch perception in a songbird. Jeffrey Cynx[a] (Department of Psychology, The Johns Hopkins University, Baltimore, MD 21218)

Six European starlings were trained to discriminate between upper harmonics of 400 Hz (2400–3600 Hz) and upper harmonics of 652 Hz (1304–3260 Hz). Analysis of the acquired discrimination showed that all birds maintained the discrimination on the basis of whether the signals contained frequencies below 2400 Hz. There was no evidence of discrimination on the basis of periodic information. The results parallel earlier work [S. H. Hulse and J. Cynx, J. Comp. Psychol. **99**, 176–196 (1985)] showing that songbirds have a bias to discriminate between frequency information on the basis of spectral cues—a kind of absolute pitch. Stimuli then were added that produced the same 400- and 652-Hz periodicities, but contained new harmonics. The starlings transferred the discrimination to these new stimuli, now showing evidence for periodicity pitch perception. Thus, although starlings (and a number of other species of songbirds) appear to be predisposed to using spectral information, they can discriminate between complex frequencies on the basis of other information if necessary. [Work supported by NIH and NSF.] [a] Present address: Rockefeller University Field Research Center, Tyrrel Road, Millbrook, NY 12545.

4:15

JJ12. Comparison of current source density analysis with multi-unit mapping in the inferior colliculus of the gerbil. David M. Harris and David C. Lambert (Department of Otolaryngology-HNS, University of Illinois College of Medicine at Chicago, Chicago, IL 60612)

In the central nucleus of the inferior colliculus single unit, 2-deoxyglucose studies, and the anatomical structure, define a laminar organization related to frequency coding. The tonotopic axis appears to be along an axis of symmetry, frequency-specific lamina are assumed to be activated synchronously, and the uniform distribution of neurons suggests homogeneous conductivity; conditions necessary for a one-dimensional current source density (CSD) analysis of the distribution of current sources and sinks evoked along the tonotopic axis. We map tone-burst-evoked multi-unit responses (multi-unit PST histograms) at 100-μ intervals as a microelectrode is lowered along the tonotopic axis. From evoked potentials, also collected at the recording sites, we compute the second derivative of the function of voltage with respect to space at a specific latency. This transformation represents the spatial profile of current sources and sinks. These data show that maximum multi-unit evoked activity has the same latency and location as the highest density of current sinks, even as latency and location shift with variations in stimulus parameters. Certain aspects do not coincide. Multi-unit data show a sustained steady-state response not seen consistently in the CSD plots and current sources do not correlate well with suppression of multi-unit activity. [Work supported by NIH.]

4:30

JJ13. A computational model for the calculation of field potentials resulting from given conductivity and current source density matrices. David C. Lambert and David M. Harris (Department of Otolaryngology-HNS, University of Illinois College of Medicine at Chicago, Chicago, IL 60612)

If a one-dimensional current source density (CSD) analysis is to be valid, certain conditions must be satisfied: (1) There must be an inherent geometric symmetry to the region being measured, and measurements must be made along an axis of symmetry. (2) Elements within the region must be activated synchronously. (3) The region must have a fairly homogeneous conductivity. We may quantify the relative influence of different factors affecting the CSD by constructing a computational model. The model is based upon the mathematical solution to the general differential equation for the calculation of the CSD. The solution gives the field potentials as a function of position in space, conductivities, and CSD distribution of the region of interest. Numerical integration methods are used to evaluate these field potentials. Once the potentials have been computed, values for the CSD along a given path through the field potential matrix can be derived and compared to the CSD distribution used to calculate the matrix. Using observations concerning the geometry and physical properties of auditory nuclei, this model will augment the construction of CSD distributions based on empirically derived data. [Work supported by NIH.]

THURSDAY AFTERNOON, 15 MAY 1986 EAST BALLROOM, 1:00 TO 4:29 P.M.

Session KK. Speech Communication VII: Speech Focus Session: Source and Tract Acoustics

Ingo R. Titze, Chairman

Department of Speech Pathology and Audiology, University of Iowa, Iowa City, Iowa 52242

Chairman's Introduction—1:00

Invited Papers

1:05

KK1. Three models of phonation. Ingo R. Titze (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Three models of phonation are described, one at the acoustic level, one at the kinematic level, and one at the biomechanic level. The models all feed the same vocal tract, which is the partial wave (Kelly–Lochbaum) transmission-line analog. The first source model is noninteractive with the vocal tract, the glottal pulse being computed by formula [Titze, J. Acoust. Soc. Am. Suppl. 1 **68**, S71 (1980)]. The second model is interactive in terms of the flow, but specifies vocal fold movement using a formula [Titze, J. Acoust. Soc. Am. **75**, 570 (1984)]. The third is interactive in terms of flow and tissue movement, i.e., it self-oscillates. Comparisons are drawn between the three models in terms of quality of sound produced, input parameters specified, and nonauditory outputs derived. Vocal tract area input to the models has recently been modified to be compatible

THURSDAY PM

with the Haskins articulatory synthesizer (companion paper in this session), so that sentence-length utterances can be produced.

1:35

KK2. An articulatory model. Thomas Baer and Philip Rubin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

In this paper, we will review the principle features of the articulatory model implemented at Haskins Laboratories [P. Rubin, T. Baer, and P. Mermelstein, J. Acoust. Soc. Am. **70**, 321–328 (1981); P. Mermelstein, J. Acoust. Soc. Am. **53**, 1070–1082 (1973)], and we will discuss its design considerations. In particular, the Haskins model is intended to identify the key articulators and their linguistically relevant control dimensions, not necessarily to accurately reproduce physiological details. Two steps in the modeling process will be focused on: the transformation from articulator "positions" to midsagital shape, and the transformation from midsagital shape to area function. We will discuss the present implementation of the model and proposed refinements. Issues to be dealt with include the degrees of freedom associated with control of the articulators, and the collection of data to improve width to cross-section rules. Design of control structures for synthesizing continuous articulations will also be discussed. [Work supported in part by NIH Grants HD 01994 and NS 13617.]

2:05

KK3. Turbulence noise sources in the vocal tract. Kenneth N. Stevens (Research Laboratory of Electronics and Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

Theoretical and experimental results relating to the generation of turbulence noise in the vocal tract are reviewed. When the turbulence occurs at the vocal-tract walls or at an obstacle, the source can be modeled as a distribution of dipole sources, which in turn can be represented in one dimension as sources of sound pressure. The source distribution is dependent on the vocal-tract configuration, so that there is interaction between the source and the mechanical configuration of the vocal tract. However, given the flow distribution for a given configuration, the acoustic properties of the airway do not appear to influence the source characteristics except under special circumstances. Calculations of spectra of the radiated sound are made for vocal-tract shapes corresponding to several aspirated and fricative consonants with various assumed distributions for the noise sources. The calculations lead to estimates of the relative level of excitation of each of the natural frequencies or formants, determined by the location and spectrum of the source. The results of the calculations are compared with measured spectra for these sounds, and from these comparisons estimates are made of the probable source distributions. [Work supported by a grant from NINCDS.]

2:35–3:05

Break

Contributed Papers

3:05

KK4. Two-mass model of the larynx: Vocal fold vibration as a negative differential resistance oscillation. William A. Conrad (30 West 71st Street #3D, New York, NY 10023) and David M. McQueen (Courant Institute of Mathematical Sciences, New York University, New York, NY 10012)

The vocal folds were analyzed as a two-mass model assuming zero glottal resistance and no supraglottal pressure recovery. Steady longitudinal flow calculations similar to those of Ishizaka and Matsudaira [SCRL Monograph No. 8 (1972)] were made, but assuming subglottal pressure varied through a sequence of steady states. A negative differential resistance was found when the dc displacement of the upper mass was small compared to that of the lower mass. Dynamic motion of the masses was represented by a pair of series resonant circuits within the glottis and transverse to the steady flow. The transverse circuits are isolated from the trachea and vocal tract by the resistances at the entrance and exit of the glottis. Sustained, self-excited, small-amplitude oscillations can be obtained when the magnitude of the negative differential resistance was equal to the real part of the circuit impedance. The oscillation frequency and phase difference between the masses depended only on myoelastic properties of the larynx. By contrast, oscillation frequency and phase in Ishizaka and Matsudaira's analysis also depends on dc aerodynamics. The calculations are therefore not contained in Ishizaka and Matsudaira's analysis. The difference in predictions should be resolved by experiment on linear controllable mechanical models.

3:17

KK5. A hybrid model of vocal fold vibration illustrating perturbations. D. Wong, M. R. Ito (Department of Electrical Engineering, University of British Columbia, Vancouver, British Columbia, V6T 1W5 Canada), and I. R. Titze (University of Iowa, Iowa City, Iowa 52242)

This study of vocal fold vibration examines the physics of vocal fold vibration and some possible causes of perturbed vibration. A multiple mass-spring computer model is used to simulate the vibratory motion of the normal vocal fold and a fold with asymmetric parameters. Speech, displacement, acceleration plots, and a mathematical discussion reducing the model to a function of lateral tissue motion are presented. Acceleration-displacement (energy exchange) diagrams from the simulation model are compared to the behavior of the mathematical equations, showing that the system driving forces can be associated with physical phenomena such as vocal tract loading, vertical phasing, Bernoulli forces, and negative damping. Simulations have produced perturbed vibrations, suggesting that 1/2 and 1/3 subharmonics are present. The 1/2 subharmonic results from vocal tract resistive and inertive loading, while the 1/3 subharmonic arises due to inertive loading and nonlinear tissue stiffness. It is necessary to derive a fourth-order system using vertical phasing of the upper and lower tissue margins, in order to drive the system into modes other than the fundamental. Irregular random perturbations also appear when parameter changes are localized.

3:29

KK6. Vocal closure patterns and source-tract acoustic interaction. Martin Rothenberg (Department of Electrical and Computer Engineering, Syracuse University, Syracuse, NY 13210)

For a strong or well-carrying voice, glottal volume velocity waveforms obtained by inverse filtering are generally consistent with the hypothesis that an interaction between the time-varying glottal impedance and an inertive component in the vocal tract impedance is causing an increase in the relative strength of higher harmonics and a decrease in average air flow. However, the required ratio of vocal tract inertance to glottal impedance is about a factor of 2 less than the greatest estimate that can be made from the glottal and vocal tract geometries using standard acoustic principles. The hypothesis that the unexplained difference is related to the shape of the vocal folds during the closing phase of their vibratory cycle was investigated by comparing glottal air flow and vocal fold contact area waveforms, with the latter obtained using an electroglottograph. To obtain an adequately accurate glottal airflow pulse and adequate synchronization of the recorded waveforms, the glottal flow waveshape was obtained by inverse filtering pharyngeal air pressure instead of the more commonly used radiated pressure or oral air flow. The transfer function of the required inverse filter is described, as well as problems and advantages of this procedure.

3:41

KK7. The stability of total phonational frequency range. Marylou Pausewang Gelfer (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Previous studies of adults [M. Cooper and N. Yanangihara, J. Commun. Disorders **3**, 261–266 (1971)] and children [M. Austin and H. Leeper, J. Commun. Disorders **8**, 309–316 (1975)] have indicated that the lowest fundamental frequency (and, presumably, range of frequencies) an individual can phonate varies by 2–4 semitones at different times of the day and from one day to another. In the present study, subjects were instructed to produce both their lowest and highest fundamental frequencies three times in one day, and three times on a second day 4–6 weeks later. Fundamental frequencies were converted to the nearest semitone, and total phonational frequency range in semitones was calculated. The stability of highest fundamental frequency, lowest fundamental frequency, and total phonational frequency range was examined both within days and between days. Results indicate that although mean variation within days is similar to previous data (approximately + / − 2 semitones), variation over the 4–6 week period is somewhat greater. Further, some individuals showed variability of six or more semitones in various measures.

3:53

KK8. Estimating subglottal pressure from esophageal pressure. Martin Rothenberg (Department of Electrical and Computer Engineering, Syracuse University, Syracuse, NY 13210) and James Mahshie (Department of Audiology, Gallaudet College, Washington, DC 20002)

A continuing problem in speech research has been the measurement of air pressure in the trachea during the speech act without invading the trachea itself. A common but problematic approach has been to use a pressure measured in the esophagus, at the level of the trachea, via a small balloon at the end of a catheter which leads to an external transducer. However, many of the problems inherent when using an external transducer can be avoided if esophageal pressure is measured using a miniature, catheter-mounted pressure transducer positioned directly in the esophagus in such a way as to have the diaphragm protected from direct contact with the esophageal walls. Recorded in this manner, esophageal pressure yields fairly accurate records of tracheal pressure, except during esophageal contraction. The frequency range available extends low enough to include the variations in average subglottal pressure that drive the various acoustic sources within the vocal tract, and high enough to include the energy at the lowest subglottal resonance. Data will be presented to support these claims and to illustrate the use of the technique in measuring the coordination between subglottal pressure changes and articulatory movements.

4:05

KK9. Accuracy of vocal tract area function estimates from acoustic reflection measurements. Paul Milenkovic (Department of Electrical and Computer Engineering, University of Wisconsin-Madison, 1415 Johnson Drive, Madison, WI 53706)

A broadband dynamic transducer is used to apply a continuous acoustic excitation at the lips for purposes of measuring the vocal tract area function. A pair of closely spaced microphones inside the tube that conveys this excitation signal is used to resolve forward and backward going acoustic wave components. The reflection coefficients of the vocal tract are obtained by a migration operator which transforms wave components measured at the lips into the equivalent causal response to a finite duration wavelet incident at successive depths into the vocal tract. The accuracy of this measurement technique is influenced by (1) the bandwidth limit imposed by the separation of the microphone pair, (2) the stability of microphone gain, and (3) the sensitivity of the microphones to structure borne vibrations. The combined influence of these limits on accuracy is determined experimentally by making acoustic measurements of the area functions of epoxy molds made to the dimensions of vowels derived from Fant's x-ray data. [Work supported by NIH grants NS 2156 and NS 16377.]

4:17

KK10. Mandibular rotation and translation during speech. Jan Edwards (Hunter College of Health Sciences, 425 East 25 Street, New York, NY 10010 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510) and Katherine S. Harris (The Graduate Center of CUNY, 33 West 42 Street, New York, NY 10036 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Jaw movement during speech has typically been represented as the movement of a single point even though this representation does not provide adequate information to predict the position of other points on the jaw. However, since the tongue rests on the jaw, such information is needed in order to calculate the jaw component of tongue position for different points on the tongue. In this experiment, a two-dimensional rigid-body model of jaw movement was developed to describe speech-related jaw opening and closing gestures. Jaw movements were decomposed into three components: rotation about the terminal hinge axis and the horizontal and vertical translation of that axis. Data were collected for three subjects during two separate recording sessions. The two-dimensional, rigid-body model of jaw movement appeared to provide useful information on the control of jaw movement. Furthermore, it proved robust enough to preserve inter-speaker differences across the two separate recording sessions. [Work supported by NINCDS grant NS-13617 to Haskins Laboratories.]

Session LL. Underwater Acoustics VI: Noise, Scattering, and Radiation

Terry E. Ewart, Chairman
Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, Washington 98105

Chairman's Introduction—1:30

Contributed Papers

1:35

LL1. Fresnel zone radiation of underwater sound created by transient moving spatially modulated patterns of laser-generated heat deposition. Hsiao-an Hsieh and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This analytical study further explores the feasibility of achieving strong farfield acoustic signals with a judicious design of the temporal and spatial pattern over which impinging laser-generated energy is added to the water surface. The heat $q(x,z,t)$ deposited per unit time and volume decays exponentially with depth z and depends on horizontal distance x and time t in such a manner that the overall configuration has Gaussian envelope factors $\exp(-t^2/T^2)$ and $\exp[-(x-Vt)^2/L^2]$, the former causing the time interval of deposition to have a duration of the order of T. A third factor, $1+\cos[k(x-Vt)]$, in conjunction with the first, causes the configuration to move as a unit with speed V in the x direction. The inhomogeneous wave equation [B-T. Chu, NACA TN 3411 (1955)] is solved by Fourier transform techniques for the case when $kL \gg 1$, $ckT \gg 1$, $kr \gg 1$, but with kL^2 not necessarily small compared with the acoustic propagation distance r. The principal analytical challenge is the evaluation of a double integral over angular frequency ω and wavenumber α; with judicious approximations we have succeeded in expressing the overall result as a linear combination of Fresnel functions of complex argument. Numerical results show a collimated beam of nearly constant frequency Vk propagating obliquely downward into the water with a gradually increasing beamwidth. [Work supported by ONR, Code 425-UA.]

1:50

LL2. Low-frequency noise fields. W. M. Carey and R. A. Wagstaff (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

Since the classic paper of Wenz [J. Acoust. Soc. Am. **34**, 1936 (1962)], ambient noise has been an extensively studied phenomena. Morris [SIO Ref. 75–34, MPL/Scripps (1975)] emphasized the importance of ships as their signals are enhanced when they cross over seamounts or proceed over the continential slopes. Wagstaff [J. Acoust. Soc. Am. **69**, 1009 (1981)] showed by the comparison of measurements and calculations that coastal shipping (ships over the continental slope and on the shelf near the slope) must be considered in order that the horizontal directionality be correctly described. He showed that these coastal sources would also affect the vertical directionality. This paper reinforces these findings with new results from down-slope transmission loss (TL), coherence, and noise directionality experiments. High-resolution noise measurements show long-term persistent directional characteristics associated with distant shipping lanes and density patterns. Short-term averages show a temporally dynamic field composed of resolved distant shipping and uncorrelated background noise. These results emphasize the importance of the coherent contribution from shipping to the mid-ocean noise field. Vertical directionality measurements by Anderson *et al.* (1972) show a broad angular distribution of noise intensity near the horizontal at low frequencies and a peaked distribution about the horizontal at high frequencies. This broad angular distribution near the horizontal was found smooth and indicates that in addition to surface ships, environmental noise influences the vertical directionality.

2:05

LL3. Oceanic ambient noise from collective oscillations of bubble clouds. A. Prosperetti (Department of Mechanical Engineering, The Johns Hopkins University, Baltimore, MD 21218)

Data show a rather substantial amount of ambient noise in the ocean in the frequency region from several hundred Hz to 1 kHz. Emission in this frequency band from isolated oscillating bubbles appears to require unrealistically large bubbles. It is proposed that the noise is produced by *bubble clouds* (caused, e.g., by wave breaking) oscillating in a collective mode. It is shown that a cloud of N equal bubbles has a lowest collective mode at a frequency of the order of $N^{-1/3}$ times the natural frequency of each bubble in isolation. A mathematical model for the description of the process is set up and preliminary results are discussed.

2:20

LL4. Oceanic noise generation by bubbles in breaking waves. Reginald D. Hollett (SACLANT ASW Research Centre, I19026 La Spezia, Italy)

A model of oceanic noise generation is described for wind-induced noise in the frequency range from a few hundred Hz to several kHz. The noise is attributed to the free oscillations of bubbles formed in wave breaking. The breaking part of the wave is treated as a uniformly moving region within which bubbles are continually generated at a constant rate. The free oscillations are excited by the excess energy imparted to the bubbles at the instant of formation. The duration of the oscillatory motion is sufficiently short that bubble extinction by buoyant rise to the surface is not considered. The bubbles are assumed to oscillate independently of their neighbors so that each bubble radiates at its natural frequency which depends on the radius. A calculation of the energy spectrum of an individual breaking wave is presented using a bubble radius distribution taken from published results of breaking wave simulations. The noise spectrum, obtained by summing over the sea surface, compares favorably with Wenz's curves.

2:35

LL5. Inverse scattering from an elastic sphere. V. M. Ayres and G. Gaunaurd (Naval Surface Weapons Center, White Oak, R-43, Silver Spring, MD 20903-5000)

The echoes (or cross sections) returned by an elastic sphere of unknown material composition can be used to determine its composition. Three parameters uniquely define the composition: the density ρ and the speeds of dilatational and shear waves, C_d, C_s. We propose an approach which capitalizes on the findings of the resonance scattering theory, which has shown how to isolate the modal resonances contained within a cross section by a process of background subtraction. These modal resonances, or equivalently the modal mechanical impedances F_n ($n = 0,1,2,...$) contain all the information required to completely and uniquely determine the composition in an asymptotic way. The problem splits naturally into two types of compositions (rubbery, or nonmetallic, and metallic), and the available cross sections can be of the monostatic or bistatic types. In all combinations of cases, the modal resonances or the modal impedances, can be used to determine the composition in an accurate way that could become the standard method for composition determination by remote sensing. Material composition can be determined as easily a the modal resonances can be extracted from the return, and we illustrate in detail how this is done in several instances.

2:50

LL6. Oscillation of a sphere in a saturated poroelastic medium using singular solutions. James J. Dlubac (Code 1905, Ship Acoustics Department, David Taylor Naval Ship R&D Center, Bethesda, MD 20084-5000) and Allen T. Chwang (Iowa Institute of Hydraulic Research, University of Iowa, Iowa City, IA 52242)

The Biot equations [J. Acoust. Soc. Am. **28**, 168–178, 179–191 (1956)] for a saturated poroelastic medium, modified to include the mass and momentum sources, are solved for the fundamental monochromatic singularities. The oscillatory point source (sourcelet) and force (forcelet) and their higher-order poles are directly applied to solve the problems of a rigid sphere embedded in an infinite medium in various modes of vibration. These closed-form solutions include the torqued and pulsating sphere, as well as the sphere in rectilinear oscillation. The sourcelet, being radially symmetric, emits only dilatational waves and represents the total volume of solid and fluid supplied. The forcelet, in addition to the two compressional waves, induces a rotational wave and is applied to the solid and fluid constituents in proportion to the porosity.

3:05

LL7. Particle velocity detection using a thin membrane. Douglas G. Todoroff and D. H. Trivett (Naval Coastal Systems Center, Panama City, FL 32407)

Underwater acoustic intensity measurements require the detection of both the acoustic pressure and the fluid particle velocity. The measurement of the acoustic pressure is easily obtained using conventional (pressure-type) hydrophones. The particle velocity may be obtained by measuring the flexural response of an appropriate membrane (constrained by an inertial structure) to the fluid particle velocity. Several thin (50–300 μm) membranes were constructed from materials which closely match (impedance, density) the acoustic medium. The use of low shear speed materials allowed for the construction of small (<10 cm) membranes which are resonant (fundamental mode) to low-frequency fluid oscillations. The response of such membranes to the acoustic particle velocity and pressure is reported in the 0.02 to 0.5 ka range.

3:20

LL8. Wave vector frequency spectrum of a transitioning boundary layer. Michel A. Josserand (Thomson-Sintra, Chemin des travails, 06802 Cagnes sur mer, France) and Gerald C. Lauchle (Applied Research Laboratory, The Pennsylvania State University, P. O. Box 30, State College, PA 16804)

The laminar-to-turbulent transition zone continues to be of interest from the flow noise point of view. Lauchle [G. C. Lauchle, J. Acoust. Soc. Am. **67**, 158–168 (1980)] has developed a theoretical model to predict the radiated component of the pressure fluctuations that occur in this zone (wavenumber less than or equal to the sonic wavenumber). In the present work, the space-time correlations of a nonhomogeneous indicator function, which monitors the existence of turbulent bursts in the transition zone, have been measured in a subsonic wind tunnel using a zero pressure gradient flat plate. An empirical model based on the measured space-time correlations has been deduced. Using existing models for the statistics of wall pressure fluctuations inside a turbulent boundary layer, the wave vector frequency spectrum of the transition zone is derived for wavenumbers greater than the sonic wavenumber. This result is of fundamental importance and may be useful in the prediction of the coupling of pressure fluctuations inside a transition zone with structural vibrations.

3:35

LL9. Measurement of radiation impedance of stepped piston radiator. Alan H. Lubell (Lubell Laboratories, Inc., 21 N. Stanwood Road, Columbus, OH 43209)

It was noted in a paper given in 1972 [A. H. Lubell, J. Acoust. Soc. Am. **52**, 1310 (1972)] that the radiation mass of a stepped piston underwater loudspeaker was approximately half of the value expected from simple piston theory. Recently, the integral equation approach was used to compute the radiation impedance for the Lubell Laboratories model 98 underwater loudspeaker and new measurements were made. This paper reviews the measurement and data reduction procedures and compares measured and theoretical radiation impedances. The original observation of reduced radiation mass is supported. A companion paper covers the integral equation computation. [This work was supported by Lubell Laboratories, Inc.]

3:50

LL10. Computation of radiation impedance of stepped piston radiator. Ralph Simon (Retired consultant, 1777 Westwood Avenue, Columbus, OH 43212)

A computer program was devised for applying the Helmholz integral formulation to determine the pressure distribution on the surface of a stepped piston radiator, which consists essentially of a series of frustrums of cones and sectors of plane circles. The same program can also be used for various axially symmetrical modes of vibration of the sphere and of the plane piston by suitably specifying the values of the input parameters that determine the shape of the surface and the angle between the normal to each surface element and its direction of vibration. Good agreement is obtained (usually $<1\%$ error) with the known analytic solutions for these latter radiators for $ka<2$, using only ten node from pole to equator. The complex radiation impedance as a function of ka was computed for Lubell Laboratories model 98 underwater loudspeaker that vibrates by separating along its flexible equator. The results are compared with those of a sphere that vibrates likewise. [This work was supported by Lubell Laboratories, Inc.]

Session MM. Biological Response to Vibration II: Effects of Low-Frequency Vibration on Performance

John C. Guignard, Chairman
824 Kent Avenue, Metairie, Louisiana 70001

Chairman's Introduction—8:30

Invited Papers

8:35

MM1. Low-frequency vibration and possible noise influence on performance. Andy W. Irwin (Department of Civil Engineering, Heriot–Watt University, Edinburgh EH14 4AS, Scotland)

Task performance in a range of situations is considered. These situations include the effects of just-perceptible vibration in quiet conditions on intricate processes to the influences of higher magnitude vibration, with or without noise intrusion, on the performance of manual dexterity tasks, reading, and in decision processes. The motion forms encompass single axis horizontal and vertical vibration, yaw vibration, and combinations of these components. Data from several laboratory studies, a wide variety of field situations, and diverse population types are described and used in the formulation of assessment methods and criteria.

9:05

MM2. Problems of defining criteria for protecting crew from ship motion effects. D. J. Thomas (1 Main Street, Chatham, NJ 07928), J. C. Guignard (824 Kent Avenue, Metairie, LA 70001), and G. J. Willems (1716 11 Street, Slidell, LA 70458)

The single-frequency model of motion sickness incidence (MSI) developed by H. F. O'Hanlon and M. E. McCauley [Aerospace Med. **45**, 366–369 (1974)] does not reliably predict MSI in complex oscillation typical of ship motion, as was shown by systematic measurements of MSI during combined heave motions [J. C. Guignard and M. E. McCauley, Aviat. Space Environ. Med. **53**, 554–563 (1982)]. Clinical and physiological observations (complemented by inertial measurements of head motion) during dynamically simulated 48-h rough-sea deployments of a projected 2000-ton surface effect ship provided further insight into the phenomenon of seasickness. Sixteen of 19 fit young naval volunteers discontinued testing, because of severe motion sickness; and all experienced symptoms. Severe motion sickness caused abandonment of psychomotor task performance; but two subjects consistently resistant to sickness performed their tasks throughout the experiment (although experiencing some nausea and exhibiting post-run postural instability). Ataxia, and symptomatic motion sickness in some cases, persisted for several hours after motion. Habituation to motion severe enough to provoke continuing vomiting did not occur in these simulations, although there was indirect evidence of some transfer of habituation from prior single-frequency motion exposure. We recommend that motion effects be studied systematically in relation to the physical parameters of provocative motion. [Work of Naval Aerospace Medical Research Laboratory Detachment #1 (now Naval Biodynamics Laboratory), New Orleans; supported by Office of Naval Research. Views not necessarily those of the Department of the Navy. Archival data in letter report to ONR (1980) and Paper #9 to NATO/DRG Seminar, Toronto (1983).]

9:35

MM3. Prediction of the incidence of motion sickness from the magnitude, frequency, and duration of vertical oscillation. A. Lawther and M. J. Griffin (Human Factors Research Unit, Institute of Sound & Vibration Research, The University, Southampton SO2 5NH, England)

A method is proposed by which the incidence of motion sickness may be predicted from measurement of the motion exposure. The method is based on data from both field and laboratory studies involving large numbers of people and is applicable to marine and other environments where vertical oscillation occurs at frequencies below 1 Hz. The frequency-dependent nature of motion sickness is incorporated by the use of a frequency weighting curve. A procedure for summing motion exposure over time yields a cumulative measure reflecting the dependency of sickness on duration of exposure. The influence of motion in axes other than the vertical is considered and the effects of population variables such as sex, age, and experience are discussed. The method enables separate predictions to be made of vomiting incidence and of reported feelings of illness.

10:05

MM4. Magnitude estimation of motion sickness in an operational environment. A. C. Bittner, Jr. (Naval Biodynamics Laboratory, Box 29407, New Orleans, LA 70189-0407) and J. C. Guignard (824 Kent Avenue, Metairie, LA 70001)

Magnitude estimation of the subjective distress of seasickness has not been used for scaling in operational environments, although it has previously been utilized in laboratory studies [e.g., O. L. Bock and C. M. Oman, Aviat. Space Environ. Med. **53**, 773–777 (1982)]. During sea trials, the method was used by one of us (ACB) to scale the provocativeness of five shipboard areas: Communications Center (CC), Communications Support Center (CSC), mess, bunk, and other areas. In addition, prediction weights for incremental changes in a "general motion illness" factor were independently derived by regression analysis on the responses of 16 crewmen during the same trials [A. C. Bittner, Jr. and J. C. Guignard, "Motion Sickness Evaluations in an At-Sea Environment: Seakeeping Trials of a USCG Cutter (WMEC 901), Naval Biodynamics Laboratory Rep. NBDL-86R002 (manuscript under review)]. These were: 0.12 (CC); 0.04 (CSC); − 0.09 (mess); − 0.35 (bunk); and 0.00 (other). Using a mess location as a comparison stimulus (assigned modulus 4), the comparable geometric means of (3–5) magnitude estimates for the same areas were: 10 (CC); 6 (CSC); 4 (mess); 2 (bunk); and 5 (other). The correlation between the direct (magnitude) and indirect (factor) scales was substantial and significant (RHO = 1.00; $p < 0.017$). These results demonstrate the utility of magnitude estimation of motion sickness for operational comparisons. [Work supported by DTNSRDC MIPR Z70099-4-00758 under an agreement with the U. S. Coast Guard. Views not necessarily those of USCG or Department of the Navy.]

10:35

MM5. Low-frequency vibration effects in the space station. Kelli F. Willshire (Mail Shop 288, NASA Langley Research Center, Hampton, VA 23665)

In response to a Presidential directive, the National Aeronautics and Space Administration (NASA) will launch a space station in the early 1990's. An objective of the Station program is to make customer use of the station cost-effective. To help meet this objective, the Space Station will be capable of permanent habitation with crew rotations of 90 days. Consequently, means of enhancing and sustaining crew productivity are receiving increased attention. Environmental factors can influence productivity either by direct effects on performance or by indirect effects which ultimately impact performance. Noise and vibration, including low-frequency vibration, are two factors being studied. This paper will discuss potential low-frequency vibration sources, impacts on Space Station crew performance, and possible solutions.

FRIDAY MORNING, 16 MAY 1986 — HASSLER ROOM, 8:15 A.M. TO 12:00 NOON

Session NN. Committee on Education in Acoustics: Strategies and Struggles for Incorporating Acoustics in Undergraduate Curricula

Robert D. Celmer, Chairman

Department of Mechanical Engineering, College of Engineering, University of Hartford, West Hartford, Connecticut 06117

Chairman's Introduction—8:15

Invited Papers

8:20

NN1. An historical review of the strategies and struggles for incorporating acoustics in the undergraduate curricula. Conrad J. Hemond and Robert D. Celmer (Department of Mechanical Engineering, College of Engineering, University of Hartford, West Hartford, CT 06117)

Beginning over 25 years ago, when the College of Engineering of the University of Hartford was in its infancy, the faculty authorized the development of an undergraduate program in Acoustics within the framework of the then Engineer's Council for Professional Development (ECPD now ABET). Hence a unique course in Engineering Acoustics was developed and offered as a requirement for the BSME degree. Problems of offering such a course were the same then as they are now. Finding a textbook with engineering applications and at an undergraduate level was and to some extent is a limiting factor. Acquisition of acoustic instrumentation and continually updating it is a limiting economic factor. Finally, securing a professor who has had some practical experience could be the greatest hurdle to overcome. This is not a necessary criterion but it is helpful to the students. This paper will trace the strategies of overcoming classical resistance to the offering of this topic to undergraduate engineers and not reserving it for a graduate program. Contrary to this idea, experience has demonstrated that the course arouses student interest and results in continuing study in graduate acoustic programs.

8:45

NN2. Undergraduate acoustics at Florida Atlantic University. A success story. J. Blaine Davidson (Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

Undergraduate acoustics has been an integral part of the Ocean Engineering program at Florida Atlantic University since its inception in 1965. Only Underwater Acoustics courses were offered in the original curriculum, but as the program matured, courses in the fundamentals of acoustics and in mechanical vibrations were added. Growth permitted more elective offerings, and in this way a course in acoustic signal processing and one in vibration, shock, and nose control were added. Today, Florida Atlantic University offers one of the strongest undergraduate acoustics programs available, and there are probably more graduates from this program entering the workplace in the field of underwater acoustics than from any other school.

9:10

NN3. Acoustics in an undergraduate engineering curriculum. David K. Holger (Department of Engineering Science and Mechanics, 3014 ME/ESM Building, Iowa State University, Ames, IA 50011)

The current status of acoustics in typical undergraduate engineering curricula, difficulties associated with increasing the exposure of undergraduate engineering students to acoustics, and strategies for doing so are discussed. Engineering undergraduates are often exposed to selected acoustical topics in an introductory physics course. Most undergraduate engineering curricula do not require formal coursework in acoustics. As a result, typical exposure to acoustical topics occurs either in elective courses that are devoted primarily to acoustics or in required courses such as vibrations or fluid mechanics where acoustics is of secondary importance. Typically, only a small percentage of engineering undergraduates select acoustics courses as electives. A practical strategy for increasing students' exposure to acoustical topics is to make appropriate elective courses available and attractive. In addition to conventional methods for making electives attractive to students and advisers, acoustics courses can be attractive from an engineering accreditation point of view. The potential for design project work, computational work, laboratory experience, and increased basic science background is present in all acoustics courses.

9:35

NN4. Strategies for building the undergraduate course sequence in architectural acoustics at the University of Florida Department of Architecture. Gary W. Siebein, Bertram Y. Kinzey, Jr., Charles F. Morgan, and Peter Burgess (Department of Architecture, 231 ARCH, University of Florida, Gainesville, FL 32611)

Over the past 10 years, the undergraduate coursework in architectural acoustics at the University of Florida Department of Architecture has evolved from a several week segment of a one semester course in Environmental Technology (which is a format typical of many schools of architecture) to a sequence where material will be covered in portions of courses in the second, third, and fourth years of study. This system was developed to respond to two basic educational needs. First, it provides the opportunity to reinforce basic concepts, hopefully increasing the students' level of comprehension. Secondly, it allows additional depth to be added in each successive year so that the level of work in the technology class closely parallels the level of work in the studio course at a given year. The process of curriculum development (struggles) used to build this course sequence as well as the development of teaching materials and methods (strategies) used in the courses will be surveyed.

10:00

NN5. Development of acoustics program at the University of Nevada, Las Vegas. Douglas D. Reynolds (Department of Civil and Mechanical Engineering, University of Nevada Las Vegas, Las Vegas, NV 89154)

The undergraduate program in Mechanical Engineering at the University of Nevada Las Vegas is less than 3 years old. The undergraduate program contains a required vibrations course and an elective noise control course. The School of Engineering at UNLV is in the process of designing a 101 000 sq ft engineering building that will house all of the engineering and computer science programs. The building will contain laboratory facilities that will include an anechoic chamber, double reverberation chambers, and a 60 000 CFM fan, and duct system connected to the larger reverberation chamber for conducting sound tests of duct elements under actual flow conditions. These facilities will be used both for demonstrations of principles associated with acoustics and with noise and vibration control and for undergraduate and graduate research projects. The author will present the procedures that were used to incorporate acoustics in the undergraduate curriculum and to develop the acoustics research facilities that will be incorporated in the new engineering building as UNLV.

10:25

NN6. Strategies and struggles for incorporating engineering acoustics in undergraduate curricula. Harold W. Lord (ME-EM Department, Michigan Technological University, Houghton, MI 49931)

The Occupational Safety and Health Act of 1970, established the principle that the employer is responsible for in-plant health and safety. The subsequent Noise Control Act of 1972 focused on the adverse effects of noise

in our work and living environment. The promulgation of these laws extended the responsibility of engineers to include the welfare of those who operate the devices and processes they design or manage, who may be exposed to unwanted emission of health related by-products. Hence, public health and safety concerns should be fundamental to engineering education and engineering acoustics should, therefore, be a logical part of the undergraduate engineering curriculum. This paper addresses the difficulties and strategies for incorporating engineering acoustics in undergraduate curricula at a time when there exists an explosive rate of new technology, and ever increasing pressures to incorporate new course material in an already over-burdened engineering curriculm. These issues are discussed in the light of recent studies and reports which deal with engineering education, namely: (1) a report by a technical panel for the National Institute of Occupational Safety and Health (1984) which gives recommendation for improving engineering practice, education, and research as it relates to occupational safety and health; (2) a report sponsored by the National Research Council (1985) which looks at the state and future of engineering education and practice in the United States; and (3) the criteria for evaluating engineering curricula set forth by the Accreditation Board for Engineering and Technology (1985–86).

10:50

NN7. Strategies and struggles of incorporating acoustics in undergraduate curriculum. Ralph W. Plummer, Terrence J. Stobbe, and James Mogensen (Department of Industrial Engineering, West Virginia University, Morgantown, WV 26506-6101)

This paper relates the philosophy of the authors concerning the introducing of integration of basic acoustical principles into junior and senior level courses. The curriculum presents basic instruction in sound measurement, the effects of noise on human performance, and acoustical control in a junior level course and reinforces these concepts plus additional instruction in acoustics in a senior level course. The paper explains the strategy and reasoning of this approach, relating both strengths and weaknesses.

Contributed Papers

11:15

NN8. Acoustics in a school of music. Douglas H. Keefe (Systematic Musicology Division, School of Music, DN-10, University of Washington, Seattle, WA 98195)

While music students are commonly exposed to musical acoustics in a formal course, there is a trend for music schools to become increasingly active in research involving acoustics, with applications to relevant subdisciplines of music. Upper-level undergraduate courses in a music school have been designed for use also by graduate students in the following areas: a full-year sequence in music science, computer applications to music, and musical applications of digital signal processing. All have instructional components involving acoustics, and all make use of a microcomputer center comprised of six music workstations. Specialized peripherals and software demonstrate acoustical phenomena and serve as tools for work in musical acoustics, music perception and cognition, music engineering, computer music, and automated music transcription. One important benefit is that effective use of computers helps music students without extensive background in calculus or the physical sciences grasp technical concepts more readily.

11:30

NN9. Waves *á la* Fourier. R. Dean Ayers (Department of Physics-Astronomy, California State University, Long Beach, CA 90840)

A senior/graduate elective course on Fourier transforms and the physics of vibrations and waves has now been taught for 6 years. The textbook by Ronald Bracewell is used in the first half of the semester to lay the mathematical foundations: piecewise functions, convolution, generalized functions, the transform and the series, theorems on transform pairs, and techniques for evaluating transforms. The second half of the semester is devoted to physical applications, with relatively little time spent on vibrations or waveforms; the major focus is on the use of spatial transforms to describe basic processes of radiation and imaging. Examples are drawn from acoustics, optics, solid-state physics, and medical imaging. The emphasis throughout this course is on the unifying structure of this approach to linear physics. Computational techniques and applications involving statistics or noise are deliberately avoided. An optional laboratory allows the students to see the ideas of the lecture course illustrated in concrete examples. Several students have gone on to do master's theses in acoustics after taking this course, and others have commented on its usefulness to them in a variety of fields.

11:45

NN10. Acoustics laboratory experiments in the undergraduate curriculum. Thomas D. Rossing (Physics Department, Northern Illinois University, DeKalb, IL 60115)

It is impossible to overemphasize the importance of laboratory experiments to the student learning acoustics. We have developed an acoustics laboratory with a library of over 50 acoustics experiments, ranging from the introductory to the advanced undergraduate/graduate level. This laboratory serves our undergraduate physics curriculum in a variety of ways, by providing: (1) experiments for a one-credit acoustics laboratory course; (2) optional experiments and "experimental problems" for an intermediate course in waves and vibration, an advanced acoustics course, and other courses; (3) open-ended experiments for undergraduate research projects; (4) demonstration experiments for lectures in mechanics, acoustics, and other intermediate courses; (5) acoustics experiments for our intermediate laboratory course; and (6) (fortunately or unfortunately) a band of well-maintained loanable test equipment for other teachers. We have prepared an acoustics laboratory manual, which is now distributed commerically by Cenco. Several experiments will be described and demonstrated as time permits.

Session OO. Engineering Acoustics IV: General Topics

David Lubman, Chairman
GM-Hughes Aircraft Company, P.O. Box 3310, Bldg. 618, MSE M415, Fullerton, California 92683

Contributed Papers

8:30

OO1. Design of high-intensity, high-efficiency sirens for acoustic agglomeration. F. G. Pla and G. Reethof (Noise Control Laboratory, The Pennsylvania State University, 213 Eng. Unit E, University Park, PA 16802)

Acoustic agglomeration requires sound pressure levels in the order of 155 to 165 dB, and powers ranging from a few hundred up to several hundred kilowatts, depending on the application. The cost of producing such power in the 1- to 4-kHz range is very significant and dictates the need for very efficient sound sources such as sirens. It is shown how nonlinear distortion of the acoustic wave and its related losses affect the performance of a siren. The presence of the mean flow through the siren passages is very important in the way it changes the local speed of sound and smoothes out the resonances of the various parts of the system. The magnitude of these effects is related to the siren operating point, i.e., the pressure ratio at which it is run and to the size of the stator ports chosen to reach the desired sound pressure level in the agglomerator. The importance of the acoustic properties of the agglomeration chamber in which the siren radiates is also critical and the chamber can be designed to minimized the losses due to nonlinear acoustic wave propagation.

8:45

OO2. Pattern recognition in acoustic signature analysis. S. Haran, R. D. Finch (Department of Mechanical Engineering University of Houston, Houston, TX 77004), and B. H. Jansen (Department of Electrical Engineering University of Houston, Houston, TX 77004)

The use of a pattern recognition approach in signature analysis is discussed. The application considered here is the inspection of railroad wheels by the analysis of acoustic signatures. Previous work [S. Haran and R. D. Finch, J. Acoust. Soc. Am. Suppl. 1 **75**, S76 (1984)] has established that the best detection method was a comparison of signatures from the two wheels on an axle. Pattern recognition techniques were used to detect the difference in the signatures of the two wheels on an axle, and to classify the wheels accordingly. After filtering and preprocessing, various features were computed from the signals for input into a pattern recognition module. Both time and frequency domain features were considered. The features include: power in various frequency bands, spectral moments, and spectral ratios. The pattern recognition package ISPAHAN was used, in which several classification schemes are available. Results from the analyses using some of the classification schemes will be presented.

9:00

OO3. The modeling of rigid-walled acoustic horns with sharp flare discontinuities. James McLean and Elmer L. Hixson (Department of Electrical and Computer Engineering, The University of Texas at Austin, TX 78712)

A composite model capable of predicting the throat impedance of so-called "constant directivity" horns is presented. These horns cannot be modeled using methods based solely on Webster's equation because they possess a sharp discontinuity which is necessary to achieve the desired radiative characteristics. The composite model combines one- and two-dimensional analyses. A two-dimensional mode-matching technique is used to model the portion of the horn in which the flare is discontinuous. The parts of the horn which flare smoothly are modeled using a generalized transmission line approach which can account for viscous losses and is shown to be equivalent to Webster's equation when losses are absent. It is necessary to use this generalized transmission line approach for the smoothly flared portions of the horn because the complex geometry of the horn prohibits analytical solution of Webster's equation. The composite model is used to predict the performance of a horn typical of the "constant directivity" horns and the predictions are compared to experimental data showing good correlation. [Work supported by JBL Inc.]

9:15

OO4. Two strategies for acoustic localization in air using a compact sensor. J. Gonzales, S. A. Deems, G. B. Netzorg, and S. L. Garrett (Department of Physics, Code 61 GX, Naval Postgraduate School, Monterey, CA 93943)

Two complementary methods for acoustic localization of an airborne sound source using a pair of microphones will be described. One approach develops a real-time solution by applying a dual time delay to the microphone outputs to form two oppositely directed cardioid beam patterns. The ratio of the difference to the sum of the cardioid signals determines the azimuthal location of the source. The second approach, which is slower, but more discriminating, uses a cross-correlation routine to find the relative time delay between the two microphone outputs. The characteristics of both methods were tested in an anechoic chamber using a synthesized helicopter signal and the helicopter signal plus an independently located white noise source. [Work supported by the Office of Naval Research.]

9:30

OO5. Finite difference errors in a two-microphone technique. A. L. Mielnicka-Pate (Department of Engineering Science & Mechanics, Iowa State University, Ames, IA 50011)

A numerical investigation of finite difference errors will be presented for a monopole and a baffled piston. The research is an extension of work reported earlier in a paper entitled "Finite Difference Nearfield Errors for a Baffled Vibrating Piston" [A. L. Mielnicka-Pate, J. Acoust. Soc. Am. Suppl. 1 **76**, S45 (1984)]. The finite difference errors were calculated for the acoustic pressure, particle velocity, sound intensity, and radiated power. The analysis included three different integration surfaces such as a plane located in a nearfield, a sphere, and a cube centered around a source. Directions and locations for significant errors in pressure, velocity, and intensity were identified. The finite difference errors will be discussed in terms of source characteristics and microphone locations. Numerical results of the radiated power calculations will be presented to illustrate the effects of a choice of integration surfaces.

9:45

OO6. Acoustic waveform and wavefront structures from pulsed planar expanding ring sources. Stephen I. Warshaw (Physics Department, University of California, Lawrence Livermore National Laboratory, P.O. Box 808, L-298, Livermore, CA 94550)

We extend previous work on analytic and semi-analytic time domain evaluations of Rayleigh's integral for the acoustic pressure field radiated

by pulsed, planar expanding ring sources imbedded in an infinite rigid baffle [S. I. Warshaw, J. Acoust. Soc. Am. Suppl. 1 77, S60 (1985) and LLNL reports UCRL-91858, 93820 (1985)] to consideration of spatial waveforms and wavefront structure of these fields. The source is taken to be a finite circular membrane having a radially symmetric, normal surface acceleration distribution $a_n = f(r)\delta(t-g(r))$. We review the quadrature techniques we use to evaluate Rayleigh's integral, and for simple choices of $f(r)$ and $g(r)$ we present theoretically calculated near and intermediate pressure fields in the form of spatial pulse profile and pressure contour snapshots at different times. We discuss the various waveform features and wavefront structures evident in these fields, as well as effects of choices of $f(r)$ and $g(r)$. The analytic methods used here can and have been implemented on a pocket computer, and have been used to benchmark a large finite-element computer code developed to simulate acoustic fields from more dynamically complex planar sources. [Work performed under the auspices of the U.S. Department of Energy by LLNL under Contract W-7405-Eng-48.]

10:00

OO7. Variational principle for harmonic fluid–structure interaction applied to an unbaffled elastic disk. J. H. Ginsberg and A. D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

A variational principle for the pressure distribution along surfaces in harmonic motion was described in an earlier paper [A. D. Pierce and X. F. Wu, J. Acoust. Soc. Am. Suppl. 1 74, S107 (1983)]. That principle was limited to situations where the particle velocity at the surface is known. Such is the case when the vibration is specified, or when an incident wave is scattered from a rigid surface. Here we shall extend the principle to treat elastic structures that are excited either by a mechanical force or by an incident wave. The principle will be illustrated by an analysis of the surface pressure and vibration pattern along a thin, circular, unbaffled plate. It will be shown that the prior formulation, which used a set of assumed modes, may be combined with a modal description of the plate vibration. The new feature of the formulation, beyond that required to consider either pressure along a rigid structure or *in-vacuo* vibration, is the need to evalute a set of coefficients that describe the coupling between the fluid and structure modes. [Work supported by the Office of Naval Research, Code 432-F.]

10:15

OO8. A potential link between structural parameters and the radiated sound field of a vibrator. Gordon Ebbitt and William Y. Strong, Jr. (CBS Technology Center, 227 High Ridge Road, Stamford, CT 06905)

Two powerful techniques available for studying structural vibration and radiation are nearfield acoustical holography (NAH) and modal analysis. The NAH technique measures the pressure field over a surface sound field in the half-space above the radiator. The modal analysis technique determines the structural parameters of the vibrator. Since both techniques can determine the velocity of the vibrating surface, it seems likely that linking these two techniques via the measured velocity could provide useful insight into how the structural parameters affect the radiated sound field. A modal analysis and a NAH analysis of a simple plate has been performed and the results of attempting to link these two techniques will be discussed.

10:30

OO9. Holographic interferometry applied to the study of in-water dynamic behavior of acoustic transducers. Howard Fein (Gould Defense Systems, Inc., Ocean Systems Division, 18901 Euclid Avenue, Cleveland, OH 44117)

Holographic interferometry has been shown to be a successful method for characterization and testing of the behavior of acoustic transducers operating in air. Characteristics of motion and displacement are known to model the ideal behavior of these devices by defining specific geometries and symmetries in time-averaged holographic interference fringe patterns [H. Fein, J. Acoust. Soc. Am. Suppl. 1 73, S25 (1983)]. These techniques have been extended to the study of the actual dynamics of acoustic transducers operating in the water medium. Successful comparisons of the differences between motion characteristics of transducers operating in water and air are shown to be of great value in defining their true dynamic behavior. Holographic data and comparisons are presented and discussed.

10:45

OO10. An algorithm for selecting transducer element array positions. King W. Wiemann and W. Jack Hughes (Graduate Program in Acoustics, Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

A lumped-parameter equivalent circuit of a tonpilz transducer is used to predict element amplitude and phase tolerance for different radiation loadings, based on in-air measurements of the transducers. Relationships among the measurable parameters of a transducer and its performance characteristics have been explored. Preliminary estimates of acceptable tolerances for each circuit parameter have been established. A two-part technique to determine the placement of transducer elements in an array which minimizes the impact of element tolerances on the directional beam pattern has been developed. This technique has been used to place, in an array, elements with tolerances of $\pm 9\%$ in amplitude and $\pm 11°$ in phase. The resulting theoretical array response achieved sidelobe levels that were within 1 dB of the -40-dB design. The first half of the technique selects four elements at a time from a larger selection pool, and places them in a manner that partially cancels their respective amplitude and phase variations. The second half of the technique uses a permutation search algorithm which rearranges the initial placement of elements in and out of the array looking for improvements in the array response.

11:00

OO11. A doubly resonant longitudinal vibrator transducer. Stephen C. Thompson (Sonar Design Group, Gould Defense Systems, Inc., Ocean Systems Division, 18901 Euclid Avenue, Cleveland, OH 44117)

The conventionally designed longitudinal vibrator transducer consisting of a double-mass-loaded active material stack is widely used in both single element and array configurations. This device, commonly called a "tonpilz" transducer, is well suited for operation at frequencies near its longitudinal resonance; however practical considerations normally limit the bandwidth attainable. It is difficult to design a tonpilz transducer with an operating mode Q of less than three which provides a limiting fractional bandwidth of approximately 0.30. This paper describes a modification to the simple tonpilz design which creates a new class of longitudinal vibrator transducer. This new device has two longitudinal vibration modes each of which can be used for operation. It is possible to design transducers of this type to be operated throughout the entire frequency band between the two resonance frequencies. Several devices of this type have been built and tested to verify that the predicted performance can be attained. Electrical and acoustical test data from these devices, which provide a flat transmitting bandwidth greater than an octave, will be presented. [Work supported by Gould Independent Research and Development Program. A patent is pending on this design type.]

11:15

OO12. Some compliance and piezoelectric properties of voided polyvinylidene fluoride (PVDF). Mark B. Moffett and James M. Powers (Naval Underwater Systems Center, New London, CT 06320)

Measurements of the compliances parallel and perpendicular to the stretched direction (s_{11} and s_{22}, respectively) as well as the piezoelectric coefficients g_{31} and g_{32} were made for Thorn-EMI voided piezoelectric polyvinylidene fluoride. The measurement method, a simple mass-spring system fed into a dual-channel spectrum analyzer, is described. Although all four measured quantities increased monotonically with increasing temperature, no glass-to-rubber transitions were encountered within the

FRIDAY AM

(− 40 to + 30) °C and (200 to 1000) Hz temperature and frequency ranges covered. [Work supported by ONR Code 220B.]

11:30

OO13. Application of diffusion bonding to thin ceramic low-voltage sonar transducers. Michael P. Johnson (Systems Engineering Department, Ocean Systems Division, Gould Defense Systems, Inc., Euclid Avenue, Cleveland, OH 44117)

Piezoelectric ceramic stacks used in sonar transducers have previously been bonded together using organic adhesives. These adhesives in general have moduli an order of magnitude lower and loss factors an order of magnitude higher than the ceramics themselves. As a consequence, degradation of the electromechanical coupling factor and decreased efficiency occur. This leads to increased operating temperatures and possibly thermal runaway. To avoid these difficulties, it has been necessary to use relatively thick ceramic to minimize the number of adhesive bonds. This results in the need for step up transformers to provide the required voltages for high power sonar. A diffusion bonding process, reported on previously [J. Acoust. Soc. Am. Suppl. 1 **72**, S23 (1982)] can be used to circumvent these difficulties and eliminate the need for step up transformers. A previously developed and tested transducer design has been modified using this process. Electrical as well as acoustical data will be presented for a 3×3 subarray of diffusion bonded elements. A comparison of effective ceramic parameters between the conventional, diffusion bonded, and modified adhesive bonded stacks will also be presented.

11:45

OO14. Continuous measurement of complex modulus from 10–30 kHz. Peter E. Madden and Timothy B. Totten (Gould, Inc., Ocean Systems Division, Cleveland, OH 44117)

The paper describes a technique which measures the complex modulus of materials in the range 10–30 kHz. The apparatus measures the input impedance of a sample of the material which is approximately 0.25 in. long and 0.7 in. in diameter. A shaker with an impedance head is used to drive one end of the sample which has a large mass at the other end. A noise drive and analysis by a dual channel spectrum analyzer connected to a desktop computer, gives a continuous frequency spectrum of the real and imaginary parts of the modulus. The method has been used over a frequency range of 10–30 kHz to measure moduli from 10^7–10^{10} Pascals with loss factors as low as 0.02 and it is felt that these ranges could be extended. The technique enables a measurement over the complete frequency range to be made in less than 5 min and the results are repeatable.

FRIDAY MORNING, 16 MAY 1986 — ALLEN ROOM, 9:00 TO 11:20 A.M.

Session PP. Musical Acoustics III

William M. Hartman, Chairman
Department of Physics, Michigan State University, East Lansing, Michigan 48824

Chairman's Introduction—9:00

Contributed Papers

9:05

PP1. Modal analysis of a handbell. 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)

Modal analysis with impact excitation offers an effective way to study the modes of vibration of percussion musical instruments [Sessions O and U, 109th ASA meeting (April 1985)]. The vibrational modes of a handbell can be conveniently classified into groups according to the number and location of their nodal circles [T. D. Rossing and R. Perrin, J. Acoust. Soc. Am. Suppl. 1 **78**, S75 (1985)]. The modes of a G_4 Malmark handbell will be described, and computer animations of modes from various groups will be shown.

9:20

PP2. Frequencies and modeshape calculations for uniform membranes. W. E. Worman (Department of Physics, Moorhead State University, Moorhead, MN 56560)

Most methods for calculating frequencies of a membrane of arbitrary shape require a reasonable guess at an approximate wavefunction. In the method presented in this paper both approximate wavefunctions and frequencies are determined. A wavefunction can always be expressed in the form

$$w(r,\theta) = \sum_{n=0}^{\infty} A_n J_n(kr) \cos n\theta + \sum_{n=1}^{\infty} B_n J_n(kr) \sin n\theta.$$

Approximate by terminating the summations at $n = N$. Require then that the boundary condition be satisfied at $2N + 1$ discrete points on the boundary to generate $2N + 1$ simultaneous homogeneous equations linear in the A's and B's. These have nontrivial solutions for the discrete k values for which the determinant of the matrix of coefficients vanishes. For one of these k values (wavenumbers) one can determine the A's and B's and hence the wavefunction. Examples of this procedure will be presented for membranes of various shapes. The method is accurate and effective requiring only a reasonable amount of time. [Work completed at Case Western Reserve University, Cleveland, OH 44106.]

9:35

PP3. A new method for measurement of drum head properties. Peter Blackstad, Martin E. Rickey (Physics Department, Indiana University, Bloomington, IN 47401), Guy St-Amant (Percussion Department, Music School, Indiana University, Bloomington, IN 47401), and Wayne R. Smith (Consulting Engineer, P. O. Box 33, Stanford, IN 47463)

For comparison of the theoretical predictions and experimental measurements of drums it is important to be able to determine the membrane tension and, in addition, any nonlinear or anisotropic properties of the membrane material. Such anisotropies can arise from the empirical methods used in the tuning of comtemporary drums in addition to the intrinsic properties of the membranes themselves. We have developed a new technique potentially applicable to a wide variety of drums. A modest and variable pressure differential is introduced into the body of the drum. This pressure is measured precisely with a water manometer. The drumhead displacement is measured with good precision in carefully selected locations and many interesting properties of the head can be determined. The results of initial measurements will be presented.

9:50

PP4. A study of air loading of timpani. Martin E. Rickey (Physics Department, Indiana University, Bloomington, IN 47401), Guy St-Amant (Percussion Department, Music School, Indiana University, Bloomington, IN 47401), and Wayne R. Smith (Consulting Engineer, P.O. Box 33, Stanford, IN 47463)

Unlike vibrating strings, which require carefully designed coupled surfaces to propagate sound, drum heads are intrinsically strongly coupled to the surrounding atmosphere. Recent studies [R. S. Christian *et al.*, J. Acoust. Soc. Am. **76**, 1336–1345 (1984)] have compared measurements made in a normal atmospheric pressure environment with theoretical analysis. We have measured time-dependent Fourier spectra of the sound output of a kettle drum at many absolute pressures between 0.01 and 1 atm. The drum was excited in a precisely controlled repeatable manner using a remote controlled mechanical striker attached to a 75-cm-diam drum. This assembly was placed in an extensively sound dampened vacuum chamber 2 m in diameter and 2 m high. The results of these measurements will be presented and compared to the previous work.

10:05

PP5. Effects of 5 years of filler and varnish seasoning on the eigenmodes in four pairs of viola plates. Carleen M. Hutchins (Catgut Acoustical Society, 112 Essex Avenue, Montclair, NJ 07042)

In 1979–1980, four pairs of top and back viola plates, finished and tuned according to Hutchins' plate tuning method, were measured for the frequency and damping of modes #1, #2, and #5 and weighted carefully. Two of the above pairs were tested with filler on, and two "in the white" with subsequent application of filler. Each pair of top and back plates was then glued to its own set of ribs and six coats of violin oil varnish applied. The four varnished "corpuses" were hung in an unheated attic for 5 years. The plates were then removed from their ribs and a similar series of tests applied. The measured changes in mode frequencies, damping, and overall weight of each plate will be presented and implications for violin making discussed.

10:20

PP6. Reflection functions of musical air columns. P. L. Hoekje and A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

Characterization of musical air columns by their input impedance $Z(f)$ has been used extensively in frequency domain discussions of the nonlinear mechanisms which sustain oscillations and has been of use to instrument makers. The McIntyre–Schumacher–Woodhouse time domain description of oscillators (1983) offers new insights into nonlinear phenomena, but requires characterization of the air columns in terms of a reflection function $r(t)$. Though $r(t)$ and $Z(f)$ are directly related, conversion by transform methods, between these parametrizations, of finite experimental data can be cumbersome or misleading. We show theoretical and experimental examples of $r(t)$ for the following terminations of a cylindrical duct: inertial, compliant, resistive, blunt open end, single side hole in infinite duct, semi-infinite tone hole lattice; these examples are compared with data for actual musical instruments. [Work supported by NSF.]

10:35

PP7. Acoustic comparison of soprano solo and choir singing. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115), Johan Sundberg, and Sten Ternström (Department of Speech Communication and Music Acoustics, Royal Institute of Technology (KTH), S-100 44 Stockholm, Sweden)

Soprano singers were recorded while singing similar texts in both choir and solo modes of performance. A comparison of long-term-average spectra of similar passages in both modes indicates that subjects used different tactics to achieve somewhat higher concentrations of energy in the 2–4 kHz range when singing in the solo mode. It is likely that this effect resulted both from a slight change of the voice source from choir to solo singing, and from a denser clustering of the higher formants, similar to that which leads to a "singer's formant" in male singers. The subjects used slightly more vibrato when singing in the solo mode. The differences observed between solo and choir singing, though not as great as previously reported in male singers [J. Acoust. Soc. Am. **76**, 541 (1984); paper submitted to J. Acoust. Soc. Am.], they are nevertheless considered to be significant. [Work supported by NSF and by the Swedish Research Council for Humanities and Social Sciences.]

10:50

PP8. Source-filter model applied to pipe organ tones. Gerald J. Lemay (Electrical and Computer Engineering Department, Southeastern Massachusetts University, North Dartmouth, MA 02747)

In the 15 years since the landmark paper by B. S. Atal and S. L. Hanauer ["Speech Analysis and Synthesis by Linear Prediction of the Speech Wave," J. Acoust. Soc. Am. **50**, 637–655 (1971)] there has been considerable interest and application of the source-filter construct for modelling the time-varying waveforms present in speech. This paper reports on the appropriateness and limitations of this model for analyzing and synthesizing different events present during the generation of musical waveforms. The first part of the paper includes a brief tutorial on the various assumptions inherent in the source-filter model. This is followed by results from analysis and synthesis experiments conducted on digitized tones from a Stuart tracker pipe organ. [Work supported by NSF Grant ECS-8408997.]

11:05

PP9. Generic spectrum envelope functions for orchestral wind instruments. A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

The basic behavior of the room-average spectrum envelopes $E(f) = M(f)T(f)$ of orchestral winds is describable via a limited number of functional forms, each with a breakpoint frequency parameter characteristic of the instrument itself. At the mezzoforte playing level the internal (mouthpiece) spectrum envelope comprises two factors, M and T_1 for the nonclarinet woodwinds and brasses. For woodwinds, $M_w(x) = \sin(\pi x)/(\pi x)^2$, where $x = f/f_{\text{break}}$, and for the brasses the coresponding function is $M_b = (1/x)(1+x^2)^{1/2}$. The clarinet odd harmonics fit $M_{co} = (1/x)/(1+x^4)^{1/2}$, while $M_{ce} = x/(1+x^8)^{1/2}$ for the evens. The function $T_1 = [(1-x)^2 + Bx^2]^{1/2}$ depends on the mouthpiece cavity-and-constriction plus main bore entrance taper. $T_1 = 1$ for clarinets. The transfer out to room-average spectra involves a tonehole based transfer function $T_h = x/(1+x^2)^{1/2}$ for nonclarinet woodwinds, and a bell-controlled function $T_b = x^{3/2}/(1+x^3)^{1/2}$ for the brasses. For the clarinet odd harmonics $T_{co} = T_h$, while $T_{ce} = 1$ for the even components. Playing-level modifications to the envelopes are similarly describable for the various instrumental types. Numerous illustrative examples will be presented. [Work supported by NSF.]

Session QQ. Speech Communication VIII: Analysis and Recognition

Thomas Baer, Chairman
Haskins Laboratories, 270 Crown Street, New Haven, Connecticut 06510

Chairman's Introduction—8:00

Contributed Papers

8:05

QQ1. Distortion analysis using Hermite polynomials. Malcolm J. Williamson (Project Phoenix, Inc., 5225 Verona Road, Madison, WI 53711-0287)

This paper describes a mathematical method for analyzing the distortion caused by a memoryless nonlinear amplifier. Two common examples of such nonlinear systems are a mu-law compander and a peak clipper. The motivation for this work is to quantify the distortion which would be present in some algorithms proposed for speech processing. The most common method for measuring distortion employs one or two sinusoids at the input and measures energy in harmonics and cross-product distortions. However, those techniques do not fully carry over to more complex signals such as speech. Our technique is to analyze the nonlinear function in terms of Hermite polynomials. This gives a measure of the total distortion as well as the distortion in each harmonic. This approach is equivalent to using Gaussian (not necessarily white) noise as a probe signal and calculating the linear component using the zero-lag cross correlation. The mathematical predictions compare closely with experimental results using short sections of speech.

8:17

QQ2. Application of smoothed Wigner distribution (WD) to speech signals. H. Garudadri, A.-P. Benguerel, J. H. V. Gilbert, and M. P. Beddoes (Audiology and Speech Sciences, University of British Columbia, Vancouver, BC V6T 1W5, Canada)

The WD is a powerful tool for the simultaneous time-frequency analysis of time-varying signals [Martin and Flandrin, IEEE Trans. Acoust. Speech Signal Process. ASSP-33 (Dec. 1985)]. For signals with simple time-frequency structures (e.g., frequency-modulated signals), the WD provides excellent resolution in the time-frequency ($t-\omega$) plane, as compared with conventional short-time spectral analysis techniques. The interpretation of the WD becomes more difficult as the time-frequency relationship of the signal gets more complex (e.g., in speech), because of the presence of cross terms and of regions below (i.e., negative) the $t-\omega$ plane. This disadvantage can be overcome by smoothing the WD in the t and ω directions over a region $T\Omega > 1$ (T and Ω are the "smear" values in time and frequency) [Janssen and Claasen, IEEE Trans. Acoust. Speech Signal Process. ASSP-33 (Aug. 1985)]. Such a smoothing process, however, makes the WD and the spectrogram equivalent in terms of the time and frequency resolutions. This presentation is directed towards comparing the performances of the WD (with partial smoothing, i.e., $T\Omega < 1$) and the spectrogram. Nonspeech signals as well as speech signals will be used to demonstrate the application of WD to the analysis of speech signals. [Work supported by NSERC, Canada.]

8:29

QQ3. Comparison of LPC estimation algorithms from noisy speech. Mohammed S. Ahmed (The Robotics Institute, Carnegie-Mellon University, Pittsburgh, PA 15213) and Asad K. Abbasi (Department of Systems Engineering, University of Petroleum & Minerals, Dhahran, Saudi Arabia)

It is well known that LPC parameters are severely distorted when the speech waveform is corrupted by noise. This paper compares the performance of different LPC estimation algorithms from noisy speech. Direct estimation of LPC parameters from noisy speech as well as the use of preprocessor speech enhancement algorithms are used. The direct estimation algorithms considered are: the autocorrelation subtraction method, shifted Yule-Walker method, and an instrumental variable method. The preprocessor speech enhancement algorithms considered are: adaptive filtering (AF), adaptive noise cancellation (ANC), and linear maximum *a priori* (LMAP) estimation. The algorithms are applied on speech corrupted by white noise. Their performance is compared through Monte-Carlo simulation conducted on voiced, unvoiced, and v/uv mixture of frames. The criteria chosen for comparison are bias, standard deviation, and average euclidean distance. The above algorithms are also applied to a speaker recognition scheme based on orthogonal LPC parameters. Comparison of the recognition results using speech corrupted by white noise is reported. Our results indicated that of the above algorithms, preprocessing by AF and ANC can improve the LPC estimation in a noisy environment. Other algorithms produce poorer results compared to the conventional autocorrelation method.

8:41

QQ4. Very low bit rate speech coding based on joint segmentation and variable length segment quantizer. Yoshinao Shiraki and Masaaki Honda (Fourth Section, Information Science Department, Electrical Communications Laboratories, NTT, 3-9-11, Midoricho, Musashino-shi, Tokyo, 180 Japan)

This paper presents a very low bit rate LPC vocoder based on a joint segmentation and quantization method using spectral segments having variable time length. The method exploits the nonuniform distribution of speech characteristics in the time and spectral domains. A measure of the spectral distance between a variable-length input speech segment and a fixed-length segment template is introduced based on linear time warping. The optimum segment boundaries and templates for a spectral sequence are efficiently determined using a dynamic programming technique so that the total spectral distortion in a voice interval is minimized. The segment templates are obtained by a sub-optimum pattern learning method, which guarantees a monotonic decrease in distortion, using a combined segmentation and clustering technique. Experimental results for a single male speaker show that this method reduces the initial distortion by 20% and yields a sound articulation score of 78%.

8:53

QQ5. Line spectral pairs—Formant correlation in speech. C. A. Wood and Thomas R. Sawallis (Motorola, Inc., 8000 W. Sunrise Boulevard, Ft. Lauderdale, FL 33322)

Researchers have stated that "closely spaced" line spectral pairs (LSP's) tend to mark formants. [J. R. Crosmer and T. P. Barnwell, *Proceedings of the ICASSP 85*, Tampa, FL, 26–29 Mar. 1985 (IEEE, New York, 1985), Vol. 2, pp. 240–243.] None of the studies quantified the term "closely spaced" and gave little or no experimental evidence to support the statement. This paper compares visually determined formant areas from digital spectrograms and overlaid LSP tracks. Incremental threshold values using three different methods for the test of "closeness" were used to test and quantify that correlation experimentally across one syllable for many talkers and across many /hVd/ syllables for a male and female talker. Results show that the closer the LSP's the more probable they are to overlay a formant, that most false hits tend to occur in low energy (i.e., consonantal) areas, and that LSP's track $F1$ and $F2$ better than $F3$ and $F4$. Incidental evidence tends to counter Crosmer and Barnwell's claim that the closed glottis odd LSP coefficient will correspond approximately to the formant center frequency.

9:05

QQ6. Calibration of computed features of isolated vowels using synthetic vowel waveforms N. B. Cox, M. R. Ito (Department of Electrical Engineering, University of British Columbia, Vancouver, BC, Canada), and M. D. Morrison (The Voice Lab, Vancouver General Hospital, Vancouver, BC, Canada)

A number of computer-extracted features of isolated vowels have been defined in the literature for use in quantification of the effects of laryngeal pathology. This paper describes the use of synthesized vowel waveforms for "calibration" of these features. The relative effects of jitter, shimmer, and additive noise perturbations on the following vowel features will be discussed; the relative average perturbation of pitch period duration and amplitude, a time domain harmonics to noise ratio [E. Yumoto, W. J. Gould, and T. Baer, J. Acoust. Soc. Am. **71**, 1544–1550 (1982)] and a frequency domain harmonics to noise ratio [H. Kojima, W. J. Gould, A. Lambiase, and N. Isshiki, Acta Otolaryngol. **89**, 547–554 (1980)]. It will be shown that the noise features are sensitive to jitter and shimmer as well as noise. A new noise feature will be introduced and shown to be effective at isolating noise perturbations from jitter and shimmer in the synthetic vowel waveforms.

9:17

QQ7. An inverse filtering technique for efficient vowel classification. Beth A. Cooper (Goodyear Aerospace Corporation, Dept. 461/C2E, 1210 Massillon Road, Akron, OH 44315) and Leon H. Sibul (Applied Research Laboratory, P.O. Box 30, State College, PA 16801)

Classification of steady-state vowel sounds by formant frequency clustering is well documented in the literature [G. E. Petersen and H. L. Barney, J. Acoust. Soc. Am. **24**, 175–184 (1952)]. A more efficient and equally effective scheme for classifying steady-state vowels has been developed using partial correlation (PARCOR) coefficients that arise as intermediate parameters in a least squares adaptive lattice inverse filter. A sixth-order autoregressive time series was synthesized from three formant/bandwidth pairs for each vowel utterance of the Petersen and Barney data. The PARCOR coefficients of the inverse filtered processes exhibited the same graphical clustering behavior as the formant frequency data, and numerical distance measures showed the two sets of parameters to be equivalent. Because the PARCOR coefficients are used directly to classify vowels and because a pth order lattice filter simultaneously generates PARCOR coefficients for all lesser order filters, PARCOR coefficients provide an efficient, minimal parametrization. These results motivate the study of this technique for processing and classifying more complex and time-varying speech signals. [Work supported by Naval Sea Systems Command.]

9:29

QQ8. Speaker-independent speech recognition using vocabulary-dependent clustering techniques. George Bernstein (GTE Laboratories Incorporated, 40 Sylvan Road, Waltham, MA 02254)

The statistical technique of template clustering for speaker-independent speech recognition have typically used the same number of clusters to represent each word in the vocabulary. However, linguistic idiosyncracies and word confusability are contributing factors in determining the number of clusters required to adequately represent a given word. Based on this notion, an algorithm has been developed to assign the number of clusters for each word automatically, using a rule based on average cluster width. The objective is to reduce the computational load without undue degradation of recognition accuracy. Experiments have been carried out with a data base of 20 male and female speakers and a 40-word vocabulary that includes the alpha digits. The range of cluster levels was 3 to 8. Thus far, experimental results have shown that, with a constant cluster density and a total of 320 centroid templates, a recognition score of 82.75% was achieved with the alpha digits. At 291 templates allocated as a variable number of centroid templates per word, a recognition accuracy of 81 percent was achieved. Additional experiments with variations of the heuristic rules will be carried out to optimize performance.

9:41

QQ9. Speaker-independent connected digit recognition with Texas Instruments data base. E. Bocchieri and G. Doddington (Speech Research, M/S 238, Texas Instruments, P.O. Box 226015, Dallas, TX 75266)

We report about speaker-independent connected-digit recognition experiments performed on Texas Instruments multi-dialect database. Our algorithm features two components. (1) A front-end "word hypothesizer" detects the possible presence (hypothesis) of vocabulary digits in the input speech. This operation uses distance measures already evaluated for isolated digit recognition ["Frame Specific Statistical Features For Speaker Independent Speech Recognition," to appear in IEEE-ASSP]. Dynamic time warping (DTW) continuously provides cumulative digit distance scores as function of input frame (time). Distance function local minima and initial/final times of relative DTW optimum paths characterize the digit hypotheses. (2) A network-based sentence recognizer selects the hypothesis sequence by minimizing a recognition score function of hypothesis distance values and initial/final times. Test and hand-labeled training data sets were collected from two distinct groups of 28 male speakers, respectively. Each set contains 7075 digit tokens in 2155 utterances. The number of digits in each utterance (from one to seven) is not provided to the recognizer. A word error rate (insertions, deletions and substitutions) of 1.1% was obtained with a ten-digit vocabulary.

9:53

QQ10. Multi-style training for robust speech recognition under stress. Richard P. Lippmann, M. M. Mack, and D. P. Paul (MIT Lincoln Laboratory, Lexington, MA 02173-0073)

A new training procedure has been developed to improve speech recognition performance when the talker is under stress and training tokens cannot be obtained in the recognition environment. Training tokens are obtained from the desired talker speaking both normally and with different talking styles (e.g., loud, soft, fast). The recognizer is then trained with all tokens and used under stress. This technique was tested with a, 35 aircraft-word vocabulary, 9 talker, 11 340 token, stress/style database. Two stress conditions included (1) workload stress created with a perceptual-motor critical-tracking task (2) intense noise presented through earphones at a 90 dB SPL level. A continuous-distribution speaker-dependent hidden Markov model recognizer was trained normally (five normally spoken tokens) and with multi-style training (one token each from normal, fast, clear, loud, and question-pitch talking styles). Error rates were 36% in noise and 12% under workload stress with normal training. Multi-style training reduced error rates substantially to 13% in noise and 9% under workload stress. Multi-style training also reduced the error rate for normally spoken words. [Supported by DARPA.]

10:05

QQ11. Multidimensional scaling of confusions produced by speech recognition systems. Moshe Yuchtman, Howard C. Nusbaum, and Chris K. Davis (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

The performance of an automatic speech recognition device is typically characterized by recognition accuracy, either in terms percentage of words recognized correctly or some measure of the types of recognition errors made by the system. As long as performance is significantly above chance or below perfect recognition, these measures cannot completely describe the operational characteristics of a recognition system. In order to provide a more complete account of recognition performance, we used multidimensional scaling of confusion matrices to compare the performance of three speech recognizers. The test vocabulary consisted of the "E" subset of the alphabet: B, C, D, E, G, P, T, V, Z. Scaling solutions were obtained for each recognizer using direct distance scores produced by the recognition algorithm, as well as distance measures (d') between each pair of words derived from confusion matrices constructed from recognition responses. The results of these analyses indicate that the patterns of errors have an underlying structure that is distinctively different for each of the three recognizers. In addition, excellent agreement exists between solutions obtained for recognizer-generated distance scores and actual recognition data. The implications of these findings for performance assessment, improved training procedures, and decision-making algorithms will be discussed. [Work supported in part by IBM.]

10:17

QQ12. Automatic recognition of isolated Spanish CV syllables.
Pedro Univaso, Enrique Rosso, and Horacio Franco (Laboratorio de Investigaciones Sensoriales, CC53 (1453), Buenos Aires, Argentina)

Due to the difficulties in the recognition of quite similar utterances, such as CV syllables with the same vowel, a two-step approach was proposed. In the first step the normalized log energy, 32 spectral band log energies, and the spectral change were used through a DP algorithm to determine: (a) one of the five broad acoustic classes of the consonant involved (voiced stop, unvoiced stop, nasal, liquid, and fricative) into which the best *matched syllable* fell and (b) the warping functions. As a second step the test pattern was compared with the reference patterns of the previous recognized class, emphasizing the differentiating regions so as to realize the *final recognition of the syllable.* The patterns were matched over the warping function taking into account only the frames around the transitional region. The final distances were calculated using only the spectral bands which focused the acoustic class distinctive features. Speaker-dependent performance over the ten more frequent Spanish CV syllables was improved from 78% to 99% with the two-step procedure instead of considering only the first step as *final recognition of the syllable.*

10:29

QQ13. Automatic recognition of intervocalic voiced stops.
Horacio Franco (Laboratorio de Investigaciones Sensoriales, CC 53 (1453), Buenos Aires, Argentina)

The recognition accuracy for intervocalic voiced plosives using a simple consonant detector combined with a context-dependent Bayesian classifier was studied. The consonantal segments were located by running a dip detector over the smoothed log-energy contour. The vocalic nuclei were located at the segments of least spectral change. LPC spectra along the vocalic segments were the features for their classification which was performed by a compound Bayes decision procedure. The feature set for the Bavesian context-dependent classification of voiced stop sounds was a sequence of three selected LPC spectra located along the amplitude dip. One spectrum was obtained at the point of minimal energy and the other two were obtained at the points of maximum slope of the smoothed log-energy contour at the beginning and end of the consonantal segment. A recognition performance of 92% was obtained in a speaker-dependent manner using a database consisting of nearly 1300 consonants embedded in VCVCVCV nonsense sequences where the V's were the vowels /a,i,u/ and the C's were the voiced stops /b,d,g/ uttered by two male Argentine Spanish speakers.