Theoretical and experimental investigation of the insertion loss of a dissipative muffler

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PROGRAM OF

The 112th Meeting of the Acoustical Society of America

Marriott Hotel • Anaheim, California • 8–12 December 1986

MONDAY EVENING, 8 DECEMBER 1986 SALONS 1 AND 2, 7:30 TO 10:00 P.M.

Tutorial Lecture

Speech production and perception. Ray Kent (Department of Communicative Disorders, University of Wisconsin, Madison, WI 53711) and Sheila Blumstein (Department of Linguistics, Brown University, Providence, RI 02912)

Generally, speech makes only conservative demands on the human capacity to generate air pressures, air flows, ranges of vocal fundamental frequency, and intensity and articulatory forces. Speaking requires a temporal regulation that may be close to the physiological limits. It has been extraordinarily difficult to define, at either the articulatory or acoustical level, those segments of speech that are uniquely and invariantly related to the presumed units of linguistic representation. Several examples of this variability problem will be given and implications will be drawn for some of the newer speech technologies. The primary challenge for speech perception is understanding how the human perceptual system converts a continuous acoustic signal into a discrete message system. A general assumption has been that the continuous acoustic signal is ultimately mapped into a discrete phonetic code relating to sound segments such as consonants and vowels. Interestingly, speech perception is not only sensitive to the variability of the signal on which the message is carried, but is also affected by the linguistic role of that message. The implication of these facts on automatic speech recognition and speech synthesis will be discussed.

TUESDAY MORNING, 9 DECEMBER 1986 SALONS 4 AND 5, 9:00 TO 11:35 A.M.

Session A. Architectural Acoustics I and Musical Acoustics I: Acoustical Evaluation of Halls for the Performing Arts, Part 1

David Lubman, Chairman

Acoustical Consultant, 14301 Middletown Lane, Westminster, California 92683

Chairman's Introduction—9:00

Invited Papers

9:05

A1. Segerstrom Hall—A review of concept, design process, and results. A. Harold Marshall (Acoustics Institute, University of Auckland, Auckland, New Zealand)

The recently opened Segerstrom Hall in Orange County, California is a 3000-seat directed reflection sequence (DRS) hall that responds in a unique asymmetrical way to the competing demands for theater and symphony. This paper reviews the origins of this concept, the design process by which it was realized, and the model study at 1:10 scale carried out in New Zealand. Comparisons between modeled results and final results in the hall will be presented.

9:35

A2. Segerstrom Hall—Evaluation of measurements and design details. Jerald R. Hyde (Consultant on Acoustics, Box 55, St. Helena, CA 94574)

The design of a large multipurpose performing space contrasts the well-known criteria of theater against the acoustical requirements for symphony and other presentations. With a fan shape of extreme width the only solution for 3000 seats, unique yet workable design solutions are required. The criteria for all uses can push against each other in ways which derive opportunities out of what first might be seen as obstacles. Segerstrom Hall is presented as an example of this creative process. The objective measurement results including early decay time (EDT), sound strength versus position, lateral energy fractions, and early-to-late energy ratios will be discussed as they relate to model study results and finally to the subjective experience of the hall itself. Details will be given of test procedures, performance mode variations of the shell and reflector design, and the acoustical curtain system.

10:05

A3.The rest of the Orange County Performing Arts Center. Dennis A. Paoletti (Paoletti/Lewitz/Associates, Inc., 40 Gold Street, San Francisco, CA 94133)

Considerable attention has been devoted in recent years at various technical meetings, including the 112th ASA Meeting, to the room acoustics design of the main theater of the Orange County Performing Arts Center in Costa Mesa, CA. In addition to the 3000-seat multi-use hall, there are four rehearsal spaces and numerous support facilities. This complex project has required a very large and thorough consulting effort in terms of sound isolation and mechanical-systems noise and vibration control in addition to room acoustics design. This paper will discuss the overall building complex, details of specific support facilities, and many interesting aspects of the consulting process that have made this project unique.

10:35

A4. Design criteria for acoustical excellence of auditoriums. Paul S. Veneklasen (Paul. S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

Based on a 25-year series of highly successful auditoriums, for which the designs drew upon a long background of musical participation and acoustical laboratory research, a summary of guidelines and quantitative achievement criteria is presented. In this paper acoustical factors are extracted from total functional requirements while recognizing that there is extensive interplay. Features are concerned with performers as well as audience. The role of laboratory modeling is stressed. The guidance role of auditorium synthesis over the years is also stressed. Adequate full-scale verification is compared with current rating schemes.

11:05

A5. Critique of certain concert-hall design criteria. A. H. Benade (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

Current concert-hall designs are often seriously unsatisfactory for performers and musically experienced listeners, who tend to pay more attention to the music itself than to the "ambiance" of a hall. Formal experiments dovetail with practical experience to support the idea that the auditory system uses early reflections to compile information on tone, pitch, spatial, and temporal location (as well as loudness) via an extension of the precedence effect. This suggests that the "early reflection" criteria with $60 < t < 80$ ms, which were originally proposed before Haas, should be based on $t < 30$ ms for at least a handful of early reflections to assure a good sampling of (among other things) the radiation patterns of the instruments. Data abound showing poor instrumental recognition when stimuli are recorded under anechoic conditions that give the listener only what the instrument happens to radiate in the direction of the microphone. However, source/listener motions and/or a few close-in reflectors (with or without instrumental onsets and decays) restore recognizability to recorded sounds. This is consistent with experiments that compare perception of earphone-presented stimuli with those arising from the statistical sound field of a room. Outlines of such experiments and of practical examples will illustrate and support the views described, and suggest the insecure basis of hall designs guided by undefined "listener preference" data obtained with variously processed versions of music recorded in an anechoic chamber. [Work supported by NSF.]

Session B. Biological Response to Vibration I and Physical Acoustics I: Physical Mechanisms of Biological Effects of Sound and Vibration, Part 1

Wesley L. Nyborg, Chairman
Department of Physics, The University of Vermont, Burlington, Vermont 05405

Invited Papers

9:00

B1. Effects of acoustic cavitation under diagnostically relevant conditions. Edwin L. Carstensen (Departments of Electrical Engineering and of Biophysics and the Rochester Center for Biomedical Ultrasound, University of Rochester, Rochester, NY 14627)

Under appropriate conditions, ultrasound is able to produce profound biological effects through its action on microscopic gas bodies in tissues. Clear examples have been found in lower organisms. To determine what this means for mammals will require a better understanding of the basic physical processes and more information about the existence of appropriate gaseous nuclei in mammalian tissues. From the limited information available at the present time, it appears that ultrasound has two qualitatively different modes of action on these bubbles. (1) Continuous wave exposures excited repetitive oscillations of the bubbles and the restraining tissue structures. It appears that this causes intercellular streaming and stresses on the cytoplasmic membrane which may be great enough to cause rupture as well as sublytic effects. (2) Ultrasound applied in very short pulses and at low temporal average intensities produces effects which have the characteristics of transient cavitation. In this case, the gas bodies expand and collapse violently producing mechanical shocks and chemical products of the extremely high temperatures in the compressed phase of the transient cavity. Although it is essential that nuclei of appropriate size be present for transient events to occur, the phenomenon of resonance which is characteristic of stable cavitation is not evident with short isolated pulses. If there are effects of cavitation in mammalian tissues under the low temporal average intensity exposures used in diagnosis, it most probably will come from the latter mechanism. Up to the present time very little effort has been directed to the study of transient cavitation under conditions which may exist within the body.

9:35

B2. Bubble formation in tissues and agar gels during ultrasonic irradiation. Stephen Daniels, Lawrence A. Crum, and Gail R. ter Haar (Oxford Hyperbaric Group, Physical Chemistry Laboratory, South Parks Road, Oxford OX1 3QZ, United Kingdom; Department of Physics, University of Mississippi, Oxford, MS 38677; and Department of Physics, Institute for Cancer Research, Sutton, Surrey SM2 5PX, United Kingdom)

Ultrasound irradiation, used in either continuous or pulsed mode, at a frequency of 0.75 MHz and spatial-temporal average intensities between 60 and 1000 mW/cm^2, has been shown to produce stable gas bubbles in the tissues of experimental animals [S. Daniels and G. R. ter Haar, Proc. Inst. Acoust. **8**, 147–154 (1986); G. R. ter Haar, S. Daniels, and K. Morton, IEEE Trans. UFFC-33, 162–164 (1986)]. Macroscopically visible bubbles are also produced in agar gels during irradiation with ultrasound at the same frequencies and intensities used with experimental animals. The effect on the number of bubbles formed of varying ultrasonic frequency, intensity, pulse length, and duty cycle as well as ambient temperature is similar in the gels to that in animals. Many of the aspects of bubble formation in gels have been explained in a qualitative manner by a theoretical model based on growth of a cavitation nucleus by rectified diffusion. This may provide a useful theoretical basis for the explanation of bubble formation *in vivo*. [Work supported by MRC (UK), NSF, ONR, NIH, and CRC/MRC Joint Committee program support for the Physics Department, Institute of Cancer Research.]

10:00

B3. *In vitro* single and multicell bioeffects studies. Morton W. Miller (Department of Biophysics, The University of Rochester, Rochester, NY 14642)

There have been two broad categories of *in vitro* bioeffects studies in relation to exposure to ultrasound, postulate testing, and empirical data gathering. Both approaches have led to results, for nonthermal exposure conditions, which are generally consistent with a mechanism involving acoustic cavitation. In general, it is thought to occur in the extracellular fluid, but some evidence is available which suggests an intracellular cavitational mechanism. This postulate needs testing. A large effort has been expended to ascertain whether ultrasound induces sister chromatid exchanges since there are a small number of reported positive effects. None of the reports has been independently confirmed, and attempts at independent confirmation have been negative. [Research supported by PHS.]

10:25

B4. The biological significance of ultrasonically induced cavitation. Mary Dyson (Division of Anatomy, United Medical and Dental Schools, Guy's Campus, London SE1 9RT, England), Lawrence A. Crum (Department of Physics, University of Mississippi, Oxford, MS 38677), and Alan Mortimer (Biomedical Engineering Section, National Research Council of Canada, Ottawa, Ontario K1A 0R8, Canada)

There is now considerable evidence that levels of ultrasound employed clinically can produce cavitation *in vitro* and, in certain circumstances, *in vivo*. Transient and prolonged cavitation is potentially damaging, destroying cells locally and inducing chemical changes through free radical production. Although not necessarily lethal, the effects of free radical activity may modify cell activity, particularly if chemical changes are produced in the components of the cell membranes. In contrast, acoustic cavitation of the stable type can have effects which may be of advantage when ultrasound is used as a therapeutic agent, although their production should be avoided in embryonic and fetal tissue; these effects include the enhancement of protein synthesis and cell motility, and could result in the stimulation of repair phenomena by ultrasound. The conditions under which cavitation occurs must be determined so that it can be avoided where advisable and employed where appropriate. [Work supported in part by the Medical Research Council and, through grants to Lawrence A. Crum, by the National Institutes of Health, National Science Foundation and the Office of Naval Research.]

10:50

B5. Ultrasonically induced cavitation in mammals. Leon A. Frizzell and Chong S. Lee (Department of Electrical and Computer Engineering, University of Illinois, 1406 W. Green Street, Urbana, IL 61801)

The role of cavitation in the production of biological effects in mammals is reviewed with emphasis on recent studies in the mouse neonate. The levels for hind limb paralysis from 1 MHz, continuous wave unfocused ultrasound in the neonatal mouse, within 24 h of birth, have been investigated at 1 and 16 bars ambient pressure and temperature between 10° and 37 °C. Above certain intensity levels the exposure duration for paralysis of 50% of specimens exposed increases with increased ambient pressure. Since an increased ambient pressure tends to suppress cavitation, such changes suggest cavitational involvement. Results show that cavitation may be involved in the resultant paralysis above approximately 150 W/cm^2 at 10 °C, and above approximately 60 W/cm^2 at 37 °C. This temperature dependence is consistent with a cavitation mechanism. [Work supported by National Institutes of Health.]

Contributed Papers

11:15

B6. Doppler ultrasound pulses can induce SCE in human lymphocytes *in vitro*. Stanley B. Barnett and Sandra M. Barnstable (Department of Biophysics, University of Rochester, 601 Elmwood Avenue, Rochester, NY 14642 and Ultrasonics Institute, Sydney NSW 2067, Australia)

Human lymphocytes were exposed to an ultrasound pulsing regimen similar to that typically employed in clinical Doppler measurements of fetal blood flow, to determine whether such pulses were able to alter the frequency of sister chromatid exchanges (SCE). A 3.1-MHz, 1.5-cm-diam transducer emitted 5- to 50-cycle pulses in a beam that was directed vertically along the axis of a stationary tube containing the blood suspension. Hydrophone measurements within the tube showed that the beam intensity profile was maintained beyond the free field focal region, thereby ensuring maximum exposure of the contents of the tube. A significant increase in SCE frequency was observed in blood from two random donors exposed to "diagnostic" levels of ultrasound, where the SPTA intensity ranged from 1 to 4 W/cm^2. However, a number of further studies on blood from a single (different) donor have failed to show any effect from ultrasound even after an exposure dwell time of 24 h. This extraordinary finding may have some relevance towards explaining some of the controversial reports of SCE induction in human lymphocytes. Research continues to try to identify the mechanisms responsible for this effect.

11:30

B7. The nature of Korotkoff signals. J. E. West (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974), S. Blank, F. B. Muller, B. Cody, G. Harshfield, J. H. Laragh, and T. G. Pickering (New York Hospital–Cornell University Medical Center, New York, NY 10003)

Analysis of Korotkoff signals (i.e., those signals generated during standard blood pressure measurements) detected under a sphygmomanometer cuff by a foil electret sensor with a frequency range extending below 0.1 Hz reveals three distinct components, K1, K2, and K3. K1 is present above systolic pressure; K2 appears at systolic and vanishes at diastolic pressure; K3 appears between systolic and diastolic pressure and continues to be present below diastolic pressure. K1 and K3 resemble the intra-arterial pressure waveform. When K3 is calibrated according to the pulse pressure, noninvasive $d(\mathrm{K3})/dt$ determinations correlated well with intra-arterial dp/dt measurements. Intra-arterial pressure recordings made with a solid-state catheter-tipped manometer distal to the cuff revealed K2 and K3 components. Comparisons of blood pressures derived from K2 with those obtained by the auscultatory (standard) method on 9 normal and 42 hypertensive subjects gave good agreement with minor systematic deviations for both systolic and diastolic values. Estimates of blood pressure using K2 were, however, closer to the intra-arterial pressure than those obtained with the auscultatory method.

11:45

B8. Measurement of the nonlinear parameter of mixtures to test Apfel's tissue composition model. Erich Carr Everbach and Robert E. Apfel (Department of Mechanical Engineering, Yale University, 2159 Yale Station, New Haven, CT 06520)

Recently a methodology was proposed for predicting tissue composition (percent water, protein, and fat) from measurements of density, sound velocity, and acoustic nonlinearity parameter [R. E. Apfel, J. Acoust. Soc. Am. **79**, 148–152 (1986)]. Comparisons of predictions with the measurements of others and with data from handbooks have been encouraging, but a more systematic comparison between theory and experiment was called for. We have established a facility for measuring accurately the sound velocity and nonlinear parameter of liquids and semiliquid substances and we are applying it to the measurement of mixtures in order to test the validity of our composition-predicting methodology. Eventually we will apply our measurement approach to tissues. Progress on experiment and theory will be presented. [Work supported in part by the Office of Naval Research and by the National Institutes of Health.]

12:00

B9. Response of polymorphonuclear leucocytes (PMNs) to vibration at kilohertz frequencies. W. L. Nyborg (Physics Department, Cook Physical Science Building, University of Vermont, Burlington VT 05405), W. L. Beeken and I. C. Northwood (Department of Medicine, Given Building, University of Vermont, Burlington, VT 05405)

It has been shown that stable volume oscillations of bubbles in a suspension of biological cells can cause significant change in the cells via small-scale manifestations of acoustic streaming, radiation force, and radiation torque. These aspects of the acoustic field are simulated, approximately, near small solid objects set into translatory vibration. Except for scale, the phenomena which have been observed at megahertz frequencies are similar to those seen at lower frequencies. In the experiments reported here suspensions of human PMNs, contained in glass capillaries, were exposed to transversely vibrating wires or strips, at frequencies of a few kilohertz, while under view through an optical microscope. As a measure of the potential of the vibrator for producing cellular change, calculations were made of the viscous stress associated with acoustic streaming near the vibrating surface. Among other findings, an increase of cell stickiness was observed at stress levels less than 1 Pa, as were changes of morphology revealed by electron microscopy. Considerable cell destruction occurred at stress levels exceeding 5–10 Pa. [Work supported in part by the NIH.]

TUESDAY MORNING, 9 DECEMBER 1986 SALON 3, 9:00 A.M. TO 12:00 NOON

Session C: Engineering Acoustics I: Transducer Models

Ilene J. Busch-Vishniac, Chairman
Department of Mechanical Engineering, The University of Texas, Austin, Texas 78712

Invited Papers

9:00

C1. Numerical models used in transducer design. Robert E. Montgomery, Clementina M. Ruggiero, and Theodore A. Henriquez (Naval Research Laboratory, Underwater Sound Reference Detachment, Orlando, FL 32806)

Numerical methods currently used to design transducers will be reviewed. The applicability, advantages, and disadvantages of each method will be discussed. Two methods, finite elements and a method for computing acoustic radiation, will be discussed in detail. These two methods will be illustrated by application to a piezoelectric Helmholtz resonator. Computed performance values such as transmitting voltage response, electrical admittance, and normal modes of vibration will be presented and compared to measured values. The presentation will conclude with an overview of needs and future plans.

9:25

C2. Innovative underwater sound transducer designs and their analysis by use of bond graphs. S. Hanish (Naval Research Laboratory, Code 5104, Washington, DC 20375-5000)

An innovative technique in modern underwater sound transducer design is to combine conventional transducers with miniature operational networks so as to achieve advanced receiving and transmitting responses for meeting newest design challenges. Operational amplifiers, field-effect transistors, and microprocessors are a few of the active or passive networks that can be so mated to underwater projectors and hydrophones. For all such electromechanical devices, the newer approach of analysis based on bond graph procedures will be demonstrated. A few examples will be analyzed in detail with the objective of both illustrating the use and advantages of bond graphs, and of unfolding the numerous possibilities for shaping transducer responses to accommodate a new generation of design demands.

9:50

C3. Bond graph modeling for modal dynamics of interacting lumped and distributed systems. Donald L. Margolis (Department of Mechanical Engineering, University of California, Davis, CA 95616)

In many practical engineering systems, it is important to understand the physics of interacting subsystems, some of which lend themselves to lumped representations while others are inherently distributed. The noise and vibration caused by the engine of an automobile is a good example. The engine, the source of the vibrational and acoustical energy, is a lumped system for typical operating speed. However, the engine sits atop the vehicle frame structure, which exhibits modal dynamics. Ultimately, the vibrational energy finds its way to surfaces that radiate acoustic energy to the surroundings. If one were to model this overall system, lumped and distributed dynamic effects would have to be included. Bond graphs are a concise pictorial representation of the energy storage, exchange, and dissipation mechanisms of dynamic engineering systems [R. C. Rosenberg and D. C. Karnopp, *Introduction to Physical System Dynamics* (McGraw-Hill, New York, 1983)]. Considerable work has been done in developing bond graph methods for the modal dynamics of distributed systems [D. L. Margolis, "Bond graphs, normal modes, and vehicular structures," Veh. Syst. Dynam. 7 (1) (1978) and D. L. Margolis, "A survey of bond graph modeling for interacting lumped and distributed systems," J. Frankin Inst. **319** (1/2) (Jan. 1985)]. The virtues of bond graph modeling are many and space is short; however, once a bond graph model has been constructed, then physical state variables are dictated, and the system can be automatically simulated using a digital computer. This applies to nonlinear as well as linear systems except that the distributed aspects of the overall system must be linear. The paper will develop bond graph modeling for lumped and distributed systems, and the procedure will be demonstrated for realistic systems.

10:15

C4. Waves-scatter bond graphs for electroacoustic systems. Henry M. Paynter (Department of Mechanical Engineering, MIT, Cambridge, MA 02139) and Ilene J. Busch-Vishniac (Department of Mechanical Engineering, University of Texas, Austin, TX 78712)

Electroacoustics was born as a twin to the telephone 110 years ago. Today electroacoustic transduction involves a sophisticated science and technology. Early in this evolution alternating currents and electrical oscillations were compared to the better-understood mechanical vibrations, but electro-technology advanced so rapidly that mechanical systems were soon treated conversely by analogous electrical circuits. Yet because a strict correspondence does not exist at the microscopic and continuum levels, several alternative analogies emerged so that now engineers confront a lack of uniform methodology, particularly when dealing with distributed transducers. Bond graphs were developed to meet just such needs by treating all physical systems as a set of multiport elements richly interconnected by another set of power bonds, which together enforce continuity-conservation laws for mass, momentum, charge, and energies. Furthermore, to provide self-consistent signal causality and detailed balances of power dissipation and entropy production, scattering variables can be employed at all ports, and wave variables on all bonds. Such characteristic variables and their methods of use were first introduced by Bernhard Riemann in 1860 for finite amplitude sound waves and later reintroduced in the 1940s for microwave circuitry. Some applications are given of this powerful approach to practical electroacoustical devices and systems.

10:40

C5. Generic models of spatially distributed sensors and actuators. Ilene J. Busch-Vishniac (Department of Mechanical Engineering, University of Texas, Austin, TX 78712)

Traditional models of sensors and actuators typically represent the transducer by a series of interconnected, discrete, lumped elements. Such models are useful because the characteristic behavior of the transducer is described by ordinary differential equations. However, these models are limited to low-frequency regimes because they contain a finite number of elements. Further, they are not easily adaptable to transducers in which the behavior is explicitly a consequence of the continuum nature of transducing element, such as acoustic horns. Presented here are generic models of transducers in which the continuous nature of the transducer is stressed. The models may be classed into two types: those which apply to transducers in which the coupling between locations occurs only through the input and output ports, and those in which there is mechanical coupling between transducer locations as well. The former case may be modeled using coupled two-port theory, and the latter using augmented transmission lines. These techniques allow for spatially varying physical parameters, and apply regardless of the transducer type, function, and geometry.

11:05

C6. Analysis of multiply coupled, multimode sonar transducer arrays. Stephen C. Thompson and Michael P. Johnson (Gould, Inc., Ocean Systems Division, Cleveland, OH 44117)

Sonar transducer element and array performance predictions using linear analysis methods have been usefully performed for over two decades. These analyses were first performed for single degree of freedom (DOF) piston transducers including only the acoustical coupling through the radiating medium. More recently, single DOF transducers including both acoustical and mechanical coupling through their common mounting surface have been analyzed [D. T. Porter, J. Acoust. Soc. Am. Suppl. 1 **78**, S73 (1985)]. However, many of the transducer types currently under consideration for use in sonar arrays are multiple DOF systems having two or more resonant modes of significance within their operating frequency band. Flextensional and flexural mode transducers, as well as several types of multimode or multiply resonant piston transducer structures, are examples of multiple DOF elements. A general method for the analysis of multiply coupled, multiple DOF

transducer arrays will be presented. The equations describing the general case have the same form as those used previously. As a consequence, many of the lessons learned from experience on arrays of single DOF elements can be extended in a straightforward manner. As an example, the concept of velocity control will be explored in different examples of multimode arrays. It will be shown that velocity control can be achieved in only a special class of multimode transducer elements—those having as many control ports (i.e., electrical) as there are structural modes.

Contributed Papers

11:30

C7. Modeling and design of transducers for electromagnetic generation of acoustics (EMAT) incorporating permeable structures. W. Imaino (IBM Almaden Research Center, San Jose, CA 95120-6099)

As a possible means for increasing the electrical to acoustical conversion efficiency as well as enhancing the spatial resolution, we have investigated the performance characteristics of transducers for electromagnetic generation of acoustics (EMATs) that incorporate permeable magnetic structures (permeable EMAT). Since the intent of this investigation is to develop high spatial resolution EMATs for noncontact scanning applications, what we term EMAT microscopy, the basic design guidelines differ somewhat from those employed in the design of more conventional EMATs, as used in pulse-echo applications, for example. A computer program has been developed, to calculate the magnetic fields, induced currents, and generated forces, of arbitrary candidate designs, taking into strict account the stray fields due to the induced currents in the structure itself, and the anisotropic polarization of the permeable structure. The results of our calculation on even very simple structures show that an enhancement of the electrical to acoustic conversion efficiency and spatial resolution can be achieved, although at the expense of operating bandwidth.

11:45

C8. A fiber-optic interferometric geophone. D. L. Gardner, R. K. Yarber, E. F. Carome, and S. L. Garrett (Physics Department, Code 61 Gx, Naval Postgraduate School, Monterey, CA 93943)

A fiber-optic interferometric geophone has been developed which consists of a seismic mass (520 g) supported by two rubber mandrels wound with a single layer of single mode optical fiber. The two mandrel-wound lengths of optical fiber, each 6.5 m long with reflecting ends, are interconnected via a fiber-to-fiber 3-dB coupler to form a Michelson interferometer. When the case of the sensor is displaced at frequencies above the mass-spring resonance frequency (i.e., in the mass-controlled frequency regime), the mass remains approximately at rest while the fiber around one mandrel is compressed and the other is expanded. This geometry has the advantage of not requiring a "reference leg" and providing four times the sensitivity of a single sensor by its "push–pull" operation and the fact that the light traverses each leg twice due to reflection. Sensitivities of 3500 rad/μ have been measured at frequencies above the mass-spring resonance, corresponding to an optical leverage factor of 700. [Work supported by the Naval Research Laboratory and the Office of Naval Research.]

TUESDAY MORNING, 9 DECEMBER 1986 SALONS C AND D, 8:30 TO 11:20 A.M.

Session D. Noise I: Aircraft Noise

Jerry J. Karlsberg, Chairman
The Boeing Company, P.O. Box 3707, Seattle, Washington 98124

Chairman's Introduction—8:30

Contributed Papers

8:35

D1. On the noise generated in the tip region of airfoils. S. A. McInerny, W. C. Meecham (MANE Department, 5732 Boelter Hall, UCLA, Los Angeles, CA 90024), and P. T. Soderman (NASA Ames Research Center, MS 247-1, Moffett Field, CA 94035)

The generation of noise by turbulence in the tip region of a blunt-tipped, lifting airfoil was studied. Data were collected on surface and farfield sound pressures in the NASA Ames 2.13×3.05 m (7×10 ft), subsonic wind tunnel using an NACA 0012 wing section of aspect ratio 2.67, at an angle of attack of 16 deg. The contributions of particular source regions were determined using cross-correlation techniques on surface and farfield sound pressures. Clipping the recorded signals prior to correlation allowed the isolation of radiated sound levels up to 25 dB below the tunnel background levels within a reasonable record length. The results indicate that in the case of blunt-tipped airfoils there is a significant noise source in addition to the separated flow under the tip vortex (on the upper wing surface) predicted by other investigators. The characteristics of these two sources of noise are presented and comparison made with existing theories and prediction schemes. [Work supported by NASA Ames Research Center.]

8:50

D2. Airfoil noise in uniform airflow. P. Garcia (Office National d'Études et de Recherches Aérospatiales, BP 72, 92322 Châtillon Cedex, France)

An experimental investigation of the NACA 0012 airfoil noise in uniform flow has been conducted in the anechoic open flow facility CEPRA 19. The purpose is to estimate the trailing edge noise, using the theoretical models developed by Chandiramani [J. Acoust. Soc. Am. **56**, 19–29 (1974)] and Howe [BBN Report No. 3679 (1977)]. These models re-

quire the knowledge of the spectral content of the turbulent-boundary-layer-induced pressure field, in the vicinity of the trailing edge. The data are obtained from longitudinal and transversal arrays of flush-mounted pressure sensors, on the airfoil. A simplified theoretical noise prediction is based on the measurement of the turbulent boundary layer pressure spectrum, the convection velocity, and the turbulence transversal integral scale. The comparison between the radiated noise measurement and the simplified prediction is reasonable in the high-frequency range only. A more general approach allows the prediction of the airfoil-radiated noise, in the whole frequency domain, and for low frequencies particularly. This general prediction formulation requires the pressure field wavenumber spectrum, induced by the turbulent boundary layer. This wavenumber spectrum is obtained from the transducer array, in contrast with Brook and Hodgson's reconstruction scheme [AIAA Paper No. 80-0977]. This approach yields a good estimate of the farfield noise. [Work supported by DRET (Direction des Recherches et Études Techniques).]

9:05

D3. Scale model investigation of propeller induced pressures on a fuselage. Werner G. Richarz (Department of Mechanical and Aeronautical Engineering, Carleton University, Ottawa, Ontario K1S 5B6, Canada)

In order to optimize sound transmission loss into the interior of an aircraft cabin, the incident pressure field must be known. Heretofore, only rms pressures were considered. It is well known that coupling with flexural waves can occur under certain conditions; thus the phase velocity of the sound field is required. For typical propeller installations, the sound field on an aircraft fuselage is governed by the nearfield of the propeller and the diffracted field. Before the sound reaches the fuselage, it must also propagate through a refracting boundary layer. In order to obtain some insight into this complex problem, a series of scale model tests have been conducted with a 7-inch propeller placed near a circular fuselage of comparable diameter. The fuselage was instrumented with ten flush-mounted microphones and could be rotated about its axis of symmetry. The rms pressures and phase velocities are measured at several blade passage frequencies and under simulated flight conditions typical of takeoff, climb, and cruise. The resultant pressure and phase contours are compared with relevant analytical models. [Work supported by the Natural Sciences and Engineering Research Council of Canada.]

9:20

D4. Efficient monitoring of aircraft noise near a military operating area. Robert W. Young and Frank T. Awbrey (Hubbs Marine Research Institute, 1720 South Shores Road, San Diego, CA 92109)

Efficient monitoring of aircraft noise near a military operating area requires: (a) adoption of A-weighted sound exposure level as the physical measure first to be applied to any noise event; (b) instrument design goals aimed exactly at the primary physical measure; (c) detailed knowledge of extreme time patterns such as sonic booms (frequency characteristics); (d) performance specifications for monitoring instruments, including tests for compliance, to ensure measurement within appropriate limits of A-weighted sound exposure level of aircraft noise characterized by extreme time patterns; (e) the procedure to apply to a succession of sound exposure levels to form 1-h average (A-weighted) sound level as the short-term summary, and to form yearly average day–night average sound level, for example, as a long-term summary; (f) tests to determine the psychoacoustic validity of time-average sound levels, both short term and long term. As an example of detailed knowledge about an extreme time pattern, this paper reports sound exposure spectrum level from 0.5 to 2000 Hz for a 100-ms sonic boom caused by fighter aircraft on Luke Air Force Range in Arizona. The C-weighted and A-weighted sound exposure levels of this and four other sonic booms measured in the same hour are utilized for illustrative calculation of A-weighted and C-weighted day–night average sound levels near a military operating area.

9:35

D5. Aircraft noise impact considerations at Orange County (JWA) Airport. Samuel R. Lane (2044 Swan Drive, Costa Mesa, CA 92626)

JWA has not been in compliance with the California Airport Noise Standard for many years. A variance process allows California airports to operate out-of-compliance and the state does not consistently or vigorously enforce the process or the regulations. A great many single-family residences are located within the excessive noise zone near the airport. The principal method used by the county to "reduce" the noise is to condemn the properties. Restrictions on commercial jet aircraft operations are also imposed that are supposed to prevent any increase in the size of the excessive noise zone. However this is a flawed process based on inconsistent reasoning that leaves an excessive noise impact unmitigated. This is because the allowable noise limit is too high, the noise zone area is inherently inaccurate, and the noise monitoring and impact rating procedures ignore significant considerations. For example, even if all commercial jet operations were terminated, the noise impact would still be excessive due to commuter and private aircraft operations.

9:50

D6. Noise reduction by damped sandwich beams. Dechang Xi and Qinghua Chen (Department of Mechanics, Zhejiang University, HangZhou, People's Republic of China)

Many papers have been published in recent years on the acousto-elasticity of damped sandwich structures. Interaction between sound fields and structural vibration is important in aerospace engineering, underwater acoustics, naval engineering, and architectural acoustics. Many authors have explored the ability of sandwich structures to increase noise reduction across a structure. Transmission loss may be improved by increasing the coincidence frequency through use of a thicker core in the sandwich. This paper describes sound-transmission-loss characteristics and corresponding structural response of a damped sandwich beam. Resonant vibration may be controlled by selection of design parameters. The concept of sound energy is used to derive a closed-form expression for transmission loss. Experimental investigations show the effects of core loss factor, sound-incidence angle, and physical and geometric parameters. Sandwich beams are shown to have better sound-transmission-loss characteristics than comparable homogenous beams.

10:05

D7. The new ANSI standard for bandpass filters. Ludwig W. Sepmeyer (1862 Comstock Avenue, Los Angeles, CA 90025) and Alan H. Marsh (DyTec Engineering, Inc., 5092 Tasman Drive, Huntington Beach, CA 92649)

The initial American standard for bandpass filters for analysis of sound or vibration was prepared by a Working Group of the Acoustical Society of America's Standards Committee S1 on Acoustics and was approved in 1966 as American National Standard ANSI S1.11-1966. The 1966 standard was re-affirmed in 1976 without technical changes. With advances in microelectronics and digital data-processing techniques available in the 1980s, it was decided in 1981 that a thorough revision of the 1966 Standard was needed. In 1986, the new ANSI Standard was published as ANSI S1.11-1986 with specifications for both analog and digital implementations of octave-band and fractional-octave-band filters including devices that utilize discrete-Fourier-transform techniques to synthesize fractional-octave-band filters. This paper describes the principal technical features of the new standard including the specification of filter design Order Number, Type Number based on error in the analysis of a white noise signal, and a Subtype designation based on error in the analysis of signals having specified positive and negative slopes. Errors are determined relative to the equivalent power that would have been transmitted by an ideal filter having infinite rejection at frequencies outside the range of specified passband-limiting frequencies. Examples are shown to demonstrate how various filter designs would conform to the specifications of the new Standard.

10:20

D8. Investigation of the dependence of excess attenuation of aircraft noise on distance. Richard K. Wolf and Richard Raspet (U.S. Army Construction Engineering Research Laboratory, Box 4005, Champaign, IL 61820-1305)

TUESDAY AM

In previous studies of sound propagation using the Fast Field Program, one aspect of our investigation has been of the excess attenuation versus a logarithmic sound velocity gradient parameter for fixed ranges [Proc. INTER-NOISE 86 Conf., 419–424 (1986)]. However, the dependence of excess attenuation upon distance is of greater interest for purposes of environmental assessment. This paper describes the decay of sound levels with distance for ground-to-ground propagation, under both upward and downward refraction conditions, and suggests how an average curve for acoustically hard and soft surfaces can be developed from this information. [Work jointly supported by the FAA and the U.S. Army Construction Engineering Research Laboratory.]

10:35

D9. Exploratory study of the potential effects of exposure to sonic boom on human health. Louis C. Sutherland (Wyle Laboratories, 128 Maryland Street, El Segundo, CA 90245) and Kenneth J. Plotkin (Wyle Laboratories, 2001 Jefferson Davis Highway, Arlington, VA 22202)

A study has been carried out to investigate possible human health effects caused by exposure of residents of the State of Nevada to sonic boom during the period from 1969–1983. The sonic boom environments were estimated from an analysis of historical Air Force records of supersonic fighter aircraft operations and on available DOD-maintained computerized records of supersonic operations of military aircraft. The estimates of sonic boom environments were considered sufficiently reliable to use in searching for a possible link to health effects. The epidemiological study on the health effects, carried out by the Department of Community and Environmental Medicine of the University of California, Irvine, involved an extensive statistical analysis of any possible correlation between sonic boom exposure and all available mortality and morbidity health data for Nevada residents for the same geographic areas and time periods. From the data collected, no convincing evidence was found to prove or disprove the existence of adverse health effects due to exposure to sonic boom. If such evidence exists, it is most likely to be found only in a prospective study of a substantial sample of individuals over time. [Work supported by U.S. Air Force AAMRL under contract with Systems Research Laboratory, Dayton, OH.]

10:50

D10. Sound producing instability of a choked jet impinging on a normal flat plate. Alan Powell (Department of Mechanical Engineering, University of Houston—University Park, Houston, TX 77004)

An axially symmetric choked air jet expands to a supersonic condition and then contracts to approximate to the sonic exit condition. This "cell" structure is repetitive in the downstream direction until destroyed by dissipation. If a rigid flat plate is introduced normal to the jet axis, a strong transverse shock wave ("normal" or "stand-off") separates the supersonic jet flow from the locally subsonic divergent flow caused by the presence of the plate. If the plate is small, the flow is unstable and emits intense sound when the plate is positioned near where a cell junction would be. For larger plates, these positions for instability merge together. The frequency then drops steadily with increasing orifice-to-plate distance, but is interrupted by upward jumps, characteristic of acoustic feedback to the orifice. Phased Schlieren photographs make clear some of the main features of the highly nonlinear oscillations of the complex supersonic/subsonic flow: In particular, the stand-off shock wave undergoes large amplitude nonlinear oscillations with consequent violent periodicity of the (mainly) subsonic flow.

11:05

D11. De-Dopplerization of aircraft flyover noise data using the WKS sampling theorem. John D. Hatlestad (Douglas Aircraft Company, 3855 Lakewood Boulevard, Mail Code 36-60, Long Beach, CA 90846)

The classical Whittaker–Kotel'nikov–Shannon (WKS) sampling theorem is applied to the de-Dopplerization of aircraft flyover noise data to remove frequency shift and smearing so that the flyover noise spectra can be more meaningfully analyzed and compared with static noise test data. The sampling theorem is used to reconstruct the uniformly sampled signal at points in time that do not correspond to sample points (i.e., to interpolate between samples). This method of interpolation is compared with one based on oversampling the signal, with linear interpolation between samples [G. P. Howell, A. J. Bradley, M. A. McCormick, and J. O. Brown, J. Sound Vib. **105**, 151–167 (1968)]. Advantages of this method include the fact that the signal can be reconstructed to any desired degree of accuracy (albeit with increasing computational load) with no special sampling considerations.

TUESDAY MORNING, 9 DECEMBER 1986 SALONS J AND K, 8:30 A.M. TO 12:00 NOON

Session E. Noise II: General Noise Topics

Gilles A. Daigle, Chairman
Division of Physics, National Research Council, Ottawa, Ontario K1A 0R6 Canada

Chairman's Introduction—8:30

Contributed Papers

8:35

E1. Theoretical and experimental investigation of the insertion loss of a dissipative muffler. Christian Pommer and Viggo Tarnow (Danish Engineering Academy, Akademivej, Building 358, 2800 Lyngby, Denmark)

By using a global theory of sound propagation in a circular duct with a dissipative material, the equivalent acoustical impedance of the cross section of the duct was calculated at low frequencies. Plane wave theory was then used to calculate the four-pole parameters of the duct using empirical formulas of the propagation constant and specific impedance of a dissipative porous material. The insertion loss of a single chamber dissipative muffler was calculated for different geometries and dissipative materials, and the insertion loss was measured using two microphone probes and white noise. The measurements were performed with different geometries and with different porous materials. The acoustical properties of the porous materials were calculated on the basis of the measured flow resistance and the calculated acoustical properties were compared with results obtained by using a standing wave tube method. A good agreement was found between the calculated and measured insertion loss of the single chamber dissipative muffler in the frequency range below the cutoff frequency of the muffler.

8:50

E2. Computer program for weapons noise propagation with terrain attenuation effects. John Wrobel, Nelson Lewis, William Russell, Robin Baldridge (U.S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5422), and Peter Guth (U.S. Military Academy, West Point, NY 10996-5000)

Whenever we are dealing with noise propagation at distances greater than 100 m from the source, there are many factors affecting the propagation. For weapons firing, a major factor affecting propagation is the tendency of terrain to diffract and deflect the sound. A computer program has been written in Turbo PASCAL to run on an IBM-XT computer. The program consists of digitized terrain data for a specific area along with a predictive sound propagation model for 25- and 7.62-mm guns along with the TOW anti-tank missile. The user selects locations of firing point, target point, and observer, along with number of rounds fired, and the start and end time of firing. The computer program provides the A-weighted day–night average sound level at the observer's location. The significant feature of NOISE.PAS is that it computes excess attenuation caused by terrain for specific sites. Additionally, NOISE.PAS takes the tedium out of doing sound calculations, which incorporate excess attenuation caused by terrain, for weapons firing.

9:05

E3. Control of noise from kidney lithotripters. J. T. Weissenburger (Engineering Dynamics International, 8420 Delmar Boulevard, Suite 303, Saint Louis, MO 63124)

The kidney lithotripter is a new medical device for removing kidney stones without requiring surgery. The patient is submerged in a tub of water and bombarded with repetitive high-energy acoustical impulses focused on the kidney. This not only disintegrates the kidney stone, but also results in significant airborne and structure-borne sound. This sound can be quite disruptive in surrounding areas if permitted to reach them. Installations are typically in existing facilities and this limits available options for noise control measures. Techniques that have been used successfully are discussed.

9:20

E4. Acoustic energy propagation in noise control foams: Approximate formulas for surface normal impedance. J. S. Bolton and E. R. Green (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

Relatively stiff, partially reticulated polyurethane foam as used in noise control applications supports two longitudinal wave types having distinct propagation factors: the "frame" wave and the "airborne" wave. Both waves appear in both the solid and porous phases of the foam. Considered here is the division of acoustic power flow (within a semi-infinite foam layer) between the two wave types and between the two phases. These divisions are strongly dependent on surface boundary conditions. Energy is carried predominantly by the frame wave and almost exclusively *via* the solid phase when the surface is sealed by an attached impermeable membrane; in this configuration foam behaves like a homogeneous elastic material having enhanced structural damping owing to the contained air. In contrast, when the surface is open, energy is carried by both wave types and is observed to oscillate between the solid and air phases. These observations have allowed the derivation of simplified formulae for the surface impedance of foam layers.

9:35

E5. The effects of microphone placement and azimuth on peak sound pressure level of type A impulses. Lawrence Shotland, Martin Robinette (Department of Speech-Language Pathology and Audiology, University of Utah, Salt Lake City, UT 84112), and Robert Brey (Communication Sciences and Disorders, Brigham Young University, Provo, UT 84602)

Single, repeatable type A impulses were generated by colliding steel spheres in an acoustically reverberant room, and measured as A-weighted sound levels by a sound level meter having four exponential time averaging modes with time constants ranging from 2 μs to 1 s. Measurements using peak hold (2-μs time constant) were then obtained on KEMAR at four azimuths and four microphone placements at each azimuth. Comparative measurements showed peak hold to give the most accurate estimation of A-weighted sound level, as verified electrically. The remaining three underestimated this value. Maximum underestimation of 34.8 dB was demonstrated using the slow time constant. KEMAR measurements demonstrated peak A-weighted sound levels to be sensitive to both microphone placement and azimuth. Measurements made in the reference field and at external microphone placements tended to underestimate peak A-weighted sound level at the tympanic membrane (TM). Two conclusions were reached. Time constant of the exponential time averaging must be shorter than the duration of the signal. Peak A-weighted sound level cannot be predicted by measurements made at sites other than at the plane of the TM.

9:50

E6. A comparative study of highway noise barrier design using two different computer models. Richard E. Burke and Krishna Nand (Engineering-Science, 75 North Fair Oaks Avenue, Pasadena, CA 91109)

Studies indicate that more people are exposed to noise from motor vehicles than any other source of noise. For this reason, the adverse noise impact of highway noise is of great concern to land use planners, regulatory agencies, and affected communities. To assess the impacts of new highway projects and to identify the effectiveness of mitigation measures, the federal government and some states require that noise analyses be conducted using specific computer models. STAMINA/OPTIMA is the Federal Highway Administration's approved model and is used by many states. SOUND 32 is the California Department of Transportation's preferred model, and is used for highway projects in California. Both models are based on similar mathematical formulations, but differ in vehicle noise emission characteristics, height of noise sources, and attenuation effectiveness for noise walls and earth berms. The purpose of this paper is to report the results of a comparative noise analysis in which both models were applied to an actual highway segment. The influence of traffic volumes, speeds, highway geometry, grade, and receptor distance on the model's estimated noise levels was studied. Differences in the location, dimensions, and effectiveness of barriers recommended by the two models are identified and some of the merits and drawbacks of the models are briefly discussed.

10:05

E7. Automated field measurements without use of an external computer for the control of instrumentation. Thor Carlsen and Gustav Ese (Norwegian Electronics a.s., P. O. Box 24, N-3408 Tranby, Norway)

Computer control over different measurement problems has in the past years become more and more popular. Computers have decreased in size and weight, while the software capacity has increased several times. However, using such a system in the field still requires that the necessary computer is brought, along with the instrumentation, to the field to make the measurements. This paper discusses a new precision sound measuring instrument (that is available also as a retrofit) with an internal BASIC software programming facility. This feature allows customized measurement sequences, calculation of final and intermittent results, control of other instruments, and graphical presentations to be carried out directly in the field on the internal video display unit without the need of an additional external computer. Descriptions of practical applications such as sound intensity, spectrum shaper control, and building acoustics measurements are also included.

10:20

E8. The use of random noise for on-line transducer modeling in an adaptive active attenuation system. L. J. Eriksson and M. C. Allie (Nelson Industries, Inc., P. O. Box 600, Stoughton, WI 53589-0600)

Active sound attenuation systems may be described using a system identification framework in which an adaptive filter is used to model the performance of an unknown acoustical plant. An error signal may be obtained from a location following an acoustical summing junction where the undesired noise is combined with the output of a secondary sound source. In order for the model output to properly converge to a value that will minimize the error signal, it is frequently necessary to determine the transfer function of the secondary sound source and the path to the error signal measurement. Since these transfer functions are continuously changing in a real system, it is desirable to perform continuous on-line modeling of the output transducer and error path. In this paper, the use of an auxiliary random noise generator for this modeling is described. Based on a Galois sequence, this technique is easy to implement, provides continuous on-line modeling, and has minimal effect on the final value of the error signal.

10:35–10:45
Break

10:45

E9. Using hearing conservation data to study the epidemiology of noise-induced hearing loss. J. Erdreich (Ostergaard Associates, Caldwell, NJ 07006) and Linda S. Erdreich (Environmental Protection Agency, Cincinnati, OH 45268)

Annual audiometric testing mandated by Federal Regulations is creating an enormous, fragmented database. Epidemiologic studies, based on this information, can potentially elucidate important parameters related to the development of noise-induced hearing loss. This paper illustrates the application of survival data analysis to the audiometric data of approximately 2500 people taken over as many as ten successive years. Survival analysis includes both the number of people who fail during a study (develop a standard threshold shift in this instance) as well as those who survive or are lost to follow-up. In this case, we found that the survival function (the probability of surviving past a time interval) and the hazard function (the probability of surviving through a time interval given that the worker has survived to the beginning of the interval) are insensitive to the exposures at the facility. However, when these functions are stratified on age at first test, older workers are at higher risk than younger workers.

11:00

E10. An acoustical shape optimization study of the nonrectangular reverberation room. Joseph R. Milner and Robert J. Bernhard (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

A computationally efficient technique for the low-frequency acoustic optimization of nonrectangular reverberation rooms is discussed. The technique utilizes the finite element shape optimization procedures discussed by Bernhard and Milner [R. J. Bernhard and J. R. Milner, "Acoustic shape optimization using the finite element method," J. Acoust. Soc. Am. Suppl. 1 **79**, S51 (1986)]. Triangular regions are shown to have properties allowing modal reductions, resulting in reduced eigensystems which afford a dramatic decrease in computational effort. By utilizing these triangular regions, the nonrectangular reverberation room may be globally optimized in an economical fashion. Results are presented using performance criteria similar to that of Louden [M. M. Louden, "Dimension ratios of rectangular rooms with good distributions of eigentones," Acustica **24** (1974)]. The character and optimal solution to the rectangular reverberation room is shown. Optimal solutions to nonrectangular rooms with various allowable room configurations are also shown.

11:15

E11. Phased array active noise controller for ducts. Werner G. Richarz (Department of Mechanical and Aeronautical Engineering, Carleton University, Ottawa, Ontario K1S 5B6, Canada)

The phased array active noise controller offers a cost-effective solution to noise control in ducts at frequencies below "cutoff." The system consists of an array of loudspeakers phased to generate a downstream propagating wave that is an inverted replicate of an incident sound wave. The signal to be controlled is sensed by a microphone or microphones upstream of the array. Each element of the array provides a fraction of the overall cancellation signal, a feature making the system a robust one. Furthermore, detrimental upstream signals arrive out of phase, and the destabilizing positive feedback paths are virtually eliminated. There are, however, installation effects, a consequence of the relative positions of the transducers in the duct. It can be shown that the bandwidth of operation as well as the maximum attenuation is governed by the relative loudspeaker–microphone separations. Analysis provides some insight and "optimum" relative positions can be found. The other installation effect is due to the radiation impedance seen by the array. Measurements of relevant impedances are presented. The overall system performance is documented for the case of a rectangular duct with and without air flow. [Work supported by the Natural Sciences and Engineering Research Council of Canada.]

11:30

E12. A blowdown water tunnel for investigation of tones from multihole plates. S. A. Elder (Physics Department, U. S. Naval Academy, Annapolis, MD 21402)

A low-noise "blowdown" water tunnel has been constructed for investigation of tones generated by multihole plates. Multihole orifices are mounted in the wall of a compact resonator chamber, simulating standing-wave conditions. The test chamber, immersed in a 260-gal open tank, is supplied by 4-in. piping connected to a 55-gal reservoir mounted on a 16-ft tower. The system is able to supply up to 80 kg/s of mass flow at a speed of 10 m/s. Water level in the test tank is kept level by means of a weir near the top, which empties into a collection barrel. Flow rate is monitored by weighing the barrel over a measured time. Results of preliminary tests will be described. [Work supported by Naval Sea Systems Command General Hydromechanics Research Program, administered by the David W. Taylor Naval Ship R & D Center, Bethesda, MD.]

11:45

E13. Utility of wind direction for predicting worst-case firing noise levels. George A. Luz, David H. Bensch (U. S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5422), Klaus Bosle, and Carl-Heinz Stokes (U. S. Army 10th Medical Laboratory, Landstuhl, West Germany)

Several German procedures for assessing the environmental noise from firing ranges have been based on the assumption that worst-case noise levels are found when the surface wind is blowing from the firing range toward the impacted community. Although there is some evidence that this assumption works for small arms ranges (e.g., 7.62-mm rifle), its utility at ranges firing larger weapons was unclear. Tests conducted with a mid-size weapon (25-mm gun) over a period of 30 days showed that the surface wind direction was not as useful a predictor as the direction of wind at 500 m above ground. With measurement sites located between 1500 and 6000 m from the firing point, the 500-m wind proved to be a fairly reliable indicator of worst case.

Session F. Psychological and Physiological Acoustics I: Speech Perception in Normal and Impaired Listeners

Judy R. Dubno, Chairman
Department of Head and Neck Surgery (Audiology), Rehabilitation Building, UCLA Medical School, Los Angeles, California 90024

Contributed Papers

9:00

F1. Frequency resolution and speech discrimination in persons with sensorineural hearing loss. Linda M. Thibodeau (Department of Speech Communication, University of Texas, Austin, TX 78712) and Dianne J. Van Tasell (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455)

Two experiments were conducted to evaluate frequency resolution and speech discrimination in a group of seven subjects with nearly equivalent flat, sensorineural hearing losses. In experiment 1, percent correct detection of a 2000-Hz tone was measured as a function of notch width in band-reject noise. In experiment 2, percent correct discrimination of synthetic speech syllables differing in spectral content in the 2000-Hz region also was evaluated as a function of notch width in the same band-reject noise. Two indices were determined from the percent correct functions obtained in each experiment: (1) the rate at which performance improved as notch width increased, and (2) the notch width at which 100% detection or discrimination occurred. Correlations among the four indices were computed. Moderate correlations were observed between tone detection and speech discrimination in notched noise. [Work supported by NIH and PHS.]

9:15

F2. Temporal processing abilities of hearing-impaired listeners. B. Espinoza-Varas and C. S. Watson (Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

It has been suggested that deficits in speech processing associated with hearing impairment may result in part from deficits in basic temporal processing abilities. Performance of 16 sensorineural hearing-impaired listeners, whose mean hearing loss ranged from 25–95 dB, was studied with a battery of discrimination tasks [Watson *et al.*, J. Acoust. Soc. Am. Suppl. 1 71, S73 (1982)]. The tasks included discrimination of: (a) duration of a 1.0-kHz tone; (b) temporal jitter within a train of square pulses; (c) temporal order of two sinusoidal frequencies; (d) temporal order within sequences of syllables; (e) detection of a variable duration test tone, embedded in a nine-tone sequence; and (f) frequency and intensity of 1.0-kHz tone. Stimuli were presented over headphones, with 6 dB/oct frequency pre-emphasis, or in the field while the subjects wore their hearing aids. The sensation level was in all cases held constant at 25–30 dB. Although there were notable individual differences, the average performance in the temporal processing tasks with tones [i.e., tasks (a), (b), (c), and (e)] was remarkably similar to that of a group of a control normal listeners ($n = 33$). However, the hearing-impaired listeners exhibited expected deficits on each of four speech processing measures (SRT) discrimination scores, syllable identification, and temporal order of syllables). No strong correlations were found between temporal processing abilities and degree of hearing loss. It therefore appears unlikely that deficits in speech processing could be explained in terms of deficits in the temporal processing abilities here studied.

9:30

F3. Discriminability of frequency-response irregularities in hearing aids. Jean A. Sullivan (Lexington Hearing and Speech Center, Jackson Heights, NY 11370), Harry Levitt (Graduate School, City University of New York, New York, NY 10036), Jain-Yih Hwang, and Ann Marie Hennessey (Lexington Hearing and Speech Center, Jackson Heights, NY 11370)

Frequency-gain characteristics of a large number of hearing aids were measured using a computer-based system for the automated measurement of hearing aids [Levitt and Sullivan, Hear. Instrum. 37, 16–18 (1986)]. Irregularities in the frequency-gain characteristic were specified in terms of the peak–valley ratio and bandwidth of each spectral peak. Typical frequency-response irregularities were then simulated on a digital master hearing aid and the detectability of both single and multiple peaks was measured using an adaptive paired-comparison technique. The average peak–valley detection threshold for single peaks was approximately 6 dB for normal-hearing subjects and 11 dB for hearing-impaired subjects. The average detection threshold for a typical set of four multiple peaks was approximately 6 dB for hearing-impaired subjects. [Research supported by NIHR.]

9:45

F4. Orthogonal polynomial compression amplification for the hearing impaired. Harry Levitt, Arlene C. Neuman, and Jayashree Toraskar (Graduate School of the City University of New York, New York, NY 10036)

The method of orthogonal polynomials is a computationally concise curve-fitting technique [Levitt and Rabiner, J. Acoust. Soc. Am. 49, 569–582 (1971)]. Each coefficient in the polynomial expansion is obtained using a single vector operation. The computational simplicity of the procedure allows for real-time speech processing using a digital master hearing aid configured around a MAP-300 array processor. The short-term spectrum of speech is first approximated by a set of orthogonal polynomials and then modified by compressing the range of variation of the polynomial coefficients. Compression of the constant term is analogous to conventional amplitude compression. Compression of the linear coefficient reduces variations in the slope of the short-term spectrum. Compression of the quadratic coefficient reduces variations in convex/concave curvature. Implementation in real time allowed for a paired-comparison adaptive strategy to be used in searching for the optimum set of compression coefficients for each subject. Compression of the constant term was found to have the largest effect leading to a significant improvement in the dynamic range of the performance-intensity function for a group of hearing impaired subjects. [Research supported by NINCDS.]

10:00

F5. Wideband compression and spectral sharpening for hearing-impaired listeners. Diane K. Bustamante and Louis D. Braida (Research Laboratory of Electronics, Room 36–749, MIT, Cambridge, MA 02139)

Normal-hearing listeners with hearing losses simulated by masking noise benefit from wideband amplitude compression under conditions (e.g. no interference, most-comfortable presentation level) for which listeners with sensorineural hearing impairments do not. Such findings suggest that speech intelligibility may be limited by degraded spectral and temporal resolution as well as reduced dynamic range for hearing-impaired listeners. To combat the combination of reduced dynamic range and reduced frequency selectivity, we have developed a speech-processing system which provides both wideband compression and a sharpening of short-term spectral shape. The sharpening is accomplished by an orthogonal decomposition of the critical-band spectrum (similar to a principal-component decomposition) followed by expansion of the component weights which specify the peak–valley structure of the spectrum. We compare speech intelligibility with this system to that with linear amplification and wideband compression alone for hearing-impaired listeners. Preliminary results for CVC nonsense syllables indicate a modest benefit for the compression/sharpening system for a male speaker, but little benefit for a female speaker. [Work supported by NIH.]

10:15

F6. The detectability and acceptability of time and frequency amplitude clipped speech. Arlene C. Neuman, Harry Levitt, and Jayashree Toraskar (Center for Research in Speech and Hearing Sciences, Graduate School, City University of New York, New York, NY 10036)

Thresholds of detectability and acceptability of unfiltered and filtered speech clipped in the time and frequency domain were obtained on normal hearing listeners. Speech signals were digitized and the clipping of the signal in the time or frequency domain was done through a digital master hearing aid. All subjects were able to detect very small amounts of clipping (less than 1%) in both the time and frequency domains. However, the intelligibility of the speech continued to be acceptable until approximately 25% of the signal was clipped. More clipping could be tolerated in the time domain than in the frequency domain. Prefiltering of the speech (high-frequency boost) allowed increased clipping in the time domain. [Research supported by NIH.]

10:30

F7. Comparison of four hearing aid prescriptive procedures. Jean A. Sullivan (Lexington Hearing and Speech Center, Jackson Heights, NY 11370), Harry Levitt (Graduate School, City University of New York, New York, NY 10036), Jain-Yih Hwang, and Ann Marie Hennessey (Lexington Hearing and Speech Center, Jackson Heights, NY 11370)

A digital master hearing aid configured around a programmable digital filter was used to simulate hearing aids prescribed by four different procedures. The prescriptive procedures were modeled on those developed by Skinner, Byrne and Tonnison, Lybarger, and Levitt *et al.* Real-time simulation allowed for adaptive paired-comparison testing to be used in evaluating the prescribed hearing aids. Data were obtained for three signal levels corresponding to input levels of 60 and 70 dB SPL and an output level corresponding to the subject's most comfortable loudness level. Speech discrimination scores were obtained for each simulated hearing aid at each signal level, as well as paired-comparison judgments of relative intelligibility and overall quality. No single prescriptive procedure was found to be uniformly superior to the other procedures considered. Differences between procedures were largest at low signal levels and smallest at the subject's most comfortable level. [Research supported by NIHR.]

10:45

F8. A log-linear approach to consonant confusion analysis. Theodore S. Bell, Donald D. Dirks (Department of Surgery, CHS 62-142, UCLA School of Medicine, Los Angeles, CA 90024), and Edward C. Carterette (Department of Psychology, University of California at Los Angeles, Los Angeles, CA 90024)

Log-linear modeling techniques, along with the likelihood-ratio chi-square (G^2) statistic, were used to assess the effects of high-frequency filtering, background noise (shaped to shift thresholds 20 dB), and presentation level on consonant confusion patterns. Specific log-linear parameters were tested to determine the significance of each of the variables and their associated interactions. Low-pass filtering changed the pattern of errors relative to unfiltered conditions as presentation level increased. This effect was evidenced by a significant interaction between stimulus, response, level, and filtering. Patterns of errors were also affected by the presence of an external noise. Signal-to-noise ratios were selected to produce equivalent performance scores relative to quiet conditions. Thus, even though the same number of errors occurred in the quiet and noise conditions, the relationship between them was significantly different. Interestingly, presentation level was also a significant variable affecting error patterns. Obviously, as presentation level increased, the number of errors decreased. However, these data indicated that the pattern of errors did not remain constant as level changed, and this effect was not similar across filtered or noise conditions. [Work supported by NINCDS.]

11:00

F9. The binaural advantage in reverberation for stimuli with controlled predictability. Stephen J. Boney (Department of Communication Disorders, University of Nebraska–Lincoln, Lincoln, NE 68583-0731) and Larry E. Humes (Department of Speech and Hearing Sciences, Indiana University, Bloomington, IN 47405)

Monaural near-ear and binaural speech-recognition performance was assessed under two levels of reverberation ($T = 0.0$ and 0.8 s), at two signal-to-babble ratios ($+4$ and -2 dB), and at two speaker-to-listener orientations (speech at 0° and noise at 180°, and speech at 45° and noise at 225°). Speech-recognition performance was assessed using the high (PH) and low (PL) predictability sentences from the revised-SPIN test (Bilger, 1984). The PH and PL items from the SPIN were recorded through a KEMAR manikin under the above conditions and played back to normal-hearing subjects via earphones. A binaural advantage was observed across all listening conditions for both the PH and PL items. The average binaural advantage for the PH items at $T = 0.0$ s was about two times greater than at $T = 0.8$ s. The binaural advantage was approximately the same for the PL items regardless of reverberation time. Significant intersubject variability was observed for the binaural advantage across listening conditions for both types of speech stimuli. In addition, there were several interaction effects between the various acoustic parameters examined for the PH and PL items.

11:15

F10. The NST in varying speech-to-noise ratios. C. E. Johnson (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740) and J. L. Danhauer (Department of Speech and Hearing Sciences, University of California, Santa Barbara, CA 93106)

The effects of a speech-spectrum noise competitor presented in varying speech-to-noise ratios on ten young adult normal-hearing listeners' responses to the nonsense syllable test (NST) were evaluated. Responses were compared to the CID W-22 test presented in the same conditions. Stimuli were presented via earphones at -10, -5, 0, $+5$, and $+10$ dB speech-to-noise ratios. Subjects' responses were scored and analyzed statistically. Results indicated: (1) performance decreased with speech-to-noise ratio; and (2) responses were poorer on the NST than on CID W-22. Clinical implications will be discussed.

11:30

F11. Spectrographic analysis of the NST—Implications for assessment and error analysis. C. E. Johnson, C. W. Asp (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740), and J. L. Danhauer (Department of Speech and Hearing Sciences, University of California, Santa Barbara, CA 93106)

Two recordings of the same word list constitute different word discrimination tests—due to talker differences and recording procedures [T. Frank and C. H. Craig, J. Speech Hear. Disorders 49, 267–271 (1984)]. Thus acoustic data of each test should be available—particularly of nonsense stimuli. A spectrographic analysis of the Edgerton–Danhauer NST was performed. Fundamental frequency F_0, formant frequencies (F_1–F_3), relative amplitude, and duration of each phoneme (when applicable) have been documented, compared across phonetic contexts, and to the results of previous research. Implications for assessment will be discussed in terms of intra/inter-speaker variability and the use of nonsense stimuli in controlled phonetic contexts. In addition, data will be used to interpret normal-hearing and hearing-impaired listeners' errors on the NST.

11:45

F12. A psychoacoustic assessment of central auditory processing in children who exhibit learning impairments. Carlton O. DeFosse (Department of Communication, University of Toledo, Toledo, OH 43606) and George B. Shirk (Department of Elementary and Early Childhood Education, University of Toledo, OH 43606)

A psychoacoustic test battery consisting of the Jack Willeford *Central Auditory Assessment Battery* and the Jack Katz *Staggered Spondaic Word* (SSW) *Test* was administered to 45 learning-impaired children that ranged in age from 6 to 16 years. Their learning impairments occurred in one or in a combination of the following academic areas: math, reading, and writing. None of the children qualified for placement in existing learning-impaired classrooms. Subtest analysis of right- and left-ear scores was performed to determine the effectiveness of the test material for identification of auditory perceptual deficits that may contribute to the children's learning impairments. Diagnostic patterns and educational implications will be discussed and interpreted.

TUESDAY MORNING, 9 DECEMBER 1986 SALONS A AND B, 8:30 A.M. TO 12:10 P.M.

Session G. Shock and Vibration I: Flow-Induced Structural Vibration

Courtney B. Burroughs, Chairman
Applied Research Laboratory, Pennsylvania State University, University Park, Pennsylvania 16802

Invited Papers

8:30

G1. A new model of wall pressure fluctuations. James M. Witting (ORI, Inc., Rockville, MD 20850)

Recent work has led to a new spectral model for pressure fluctuations at a rigid plane wall bounding a turbulent boundary layer, based on an explicit representation of structures in the boundary layer near the wall—bursts and sweeps—that exchange momentum within the layer and support the shear stress there [J. M.Witting, Noise Control. Eng. J. 26, 28–43 (1986)]. The fluctuating wall pressure results from the contributions of a collection of independent bursts and sweeps. Each burst and sweep is modeled as a dipole flow that moves with the local mean flow, has a finite duration, and has the correct strength to mix the fluid through a Prandtl mixing length above and below its center. This paper describes some of the salient features and extensions of the model, emphasizing (1) the physical interpretation of the model structure and its three adjustable parameters, which are sufficient to yield predictions for all frequencies and wavelengths, (2) the ranges of frequencies and wavenumbers over which the model can be applied with confidence, based both on theory and past experiments, and (3) the applicability of the modeling approach to other situations.

8:55

G2. Boundary layer transition as a source of noise and vibration. Gerald C. Lauchle (Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804) and M. A. Josserand (Thomson–Sintra, Chemin des travails, 06802 Cagnes sur mer, France)

When laminar flow over a rigid or flexible surface becomes unstable, an intermittent flow state occurs. This intermittent flow regime, called the transition region, is where turbulent spots are created, and then grow as they convect downstream at a velocity typically equal to 0.7 times the free-stream velocity. The spots eventually coalesce to form the beginning of the fully developed turbulent boundary layer. The statistics of the velocity or pressure fluctuations in the transition region are essentially stationary in time, but nonhomogeneous in the streamwise direction. Fundamentally, it has been argued that this region is capable of creating monopole sound radiation, e.g., Lauchle [J. Acoust. Soc. Am. 69, 665–671 (1981)] and Sornette and Lagier [Acustica 55, 255–267 (1984)]. Also, it has been suspected that a transitional boundary layer can induce wall vibrations. These issues have been under study for some time. We have completed a set of measurements on the space-time statistics of turbulent spots in a naturally occurring transition zone and from them developed an analytical model for the wavenumber-frequency spectrum of the pressure fluctuations. Based on this model, it appears that the transition zone wall pressure is less intense than that of a fully developed turbulent layer by a factor equal approximately to the intermittency factor. This presentation will review the current research findings on wall pressure fluctuations and radiated sound caused by boundary layer transition. [Work supported by Applied Research Laboratory under NAVSEA contract.]

9:20

G3. Wave vector-frequency spectra of nonhomogeneous fields. Wayne A. Strawderman (Naval Underwater Systems Center, New London Laboratory, New London, CT 06320)

The utility of wave vector-frequency spectral analysis for the description and interpretation of hydroacoustic and structural-acoustic fields has been amply demonstrated over the past decade in both theoretical and experimental applications. In the majority of these applications, the statistics of the random fields of interest were assumed to be stationary and homogeneous. While many of the hydroacoustic and structural-acoustic fields of practical interest can be considered stationary, few can be considered homogeneous. Structural-acoustic fields are nonhomogeneous owing to the space-varying nature (e.g., boundaries) of practical structures. In hydroacoustics, the natural growth of the turbulent boundary layer results in a nonhomogeneous pressure field at the boundary. This paper addresses the application of wave vector-frequency analysis to the description and interpretation of nonhomogeneous, but stationary, fields. Whereas only one definition of the wave vector-frequency spectrum exists for a homogeneous, stationary field, several alternative definitions of the wave vector-frequency spectrum are possible for the nonhomogeneous, stationary field. The utility of these various spectral forms for the analysis and interpretation of nonhomogeneous, stationary fields is assessed.

9:45

G4. Analytical structural acoustics of submerged cylindrical shells. A. Harari and B. E. Sandman (Naval Underwater Systems Center, Newport, RI 02841)

A detailed analytical model of ring-stiffened cylindrical shell vibration is presented. The model is to accept excitation due to turbulent boundary layer pressure fluctuations. The mathematical model is capable of predicting the acoustical effects of a compliant decoupling layer attached to the external surface of the shell structure. Both the shell vibration levels and farfield acoustic pressures are predicted. A few focused example results are presented with and without acoustical decouplers.

Contributed Papers

10:10

G5. Wall pressure measurements and acoustics in turbulent pipe flow. Mark A. Daniels and Gerald C. Lauchle (Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, PA 16804)

Measurements of the turbulent boundary layer (TBL) wall pressure spectrum and acoustic field were conducted in the Boundary Layer Research Facility of the Applied Research Laboratory. This facility uses glycerine as the working fluid. Subminiature, piezoresistive pressure transducers were used for these measurements. The TBL wall pressure spectrum was obtained using a novel signal processing technique (transducer difference signals) that minimized both acoustic and vibration-induced noise while maintaining the integrity of the measured TBL wall pressure spectrum. A measurement involving the coherence function between these transducer difference signals validated the measured TBL wall pressure spectra and all assumptions used in the development of the measurement technique. The measured nondimensionalized spectra of the TBL fluctuating wall pressure are compared to those measured in previous investigations. These comparisons have substantiated a maximum, normalized transducer diameter for the complete resolution of the high-frequency spectral energy associated with the pressure fluctuations within the TBL. In this investigation, a transducer diameter of 2.1 viscous wall units is demonstrated. This is a factor of 9, smaller than ever before achieved. [Work supported by Applied Research Laboratory E/F Program under NAVSEA contract.]

10:25

G6. Fluid loading on vibrating plates in a uniform flow field via a wave-vector/time domain method. Dong-Jye Li and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

A new approach is presented to evaluate the fluid loading on vibrating elastic plates with specified time-dependent velocities in uniform flows. An *in vacuo* eigenfunction expansion with time-dependent coefficients is used to describe the specified normal velocity of the plate. Acoustical fluid loading on the plate is expressed as an eigenfunction expansion in which each modal coefficient is expressed as a summation of convolution integrals involving fluid/modal impulse responses and the modal velocities coefficients. Wave-vector/time domain methods are used to develop expressions for the fluid modal impulse responses. Numerical results are presented to illustrate the characteristics of the fluid/modal impulse responses and fluid loading for various plates and Mach numbers. (Work supported by ONR.]

10:40

G7. Planar elastic vibrators in a uniform flow field with broadband mechanical excitations. Dong-Jye Li and Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

Flow-induced forces on structures can significantly affect their dynamic response. A new method is presented to evaluate the dynamic response of planar elastic vibrators in uniform fluid flow fields to broadband mechanical excitations. The approach is based on the use of *in vacuo* eigenfunction expansion to solve the fluid-loaded problem. The time-dependent coefficients of the modal expansion for the velocity are shown to satisfy a set of coupled convolution integral equations which involve mode and Mach number-dependent impulse responses. Pressures in the field are simply expressed as a summation of uncoupled convolution integrals. Numerical results are presented to illustrate the effects of Mach number on the transient response of a plate subjected to a broadband mechanical excitation. [Work supported by ONR.]

10:55

G8. Interaction between a laminar boundary layer and an elastic layer. Mauro Pierucci and Pedro Morales (Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

A two-dimensional viscous flow field is bounded on both sides by a purely elastic layer of finite thickness that supports both shear and longitudinal waves. The mean flow field is assumed to be a Poiseuille flow. The boundary layer equations are linearized to the Orr–Sommerfield equation and the interaction between the fluid and the elastic layer occurs through the continuity of velocity and stress at the fluid–solid interface. The stability of the disturbances in the fluid and the elastic layer is analyzed. Kaplan ["The Stability of a Laminar Incompressible Boundary Layer in the Presence of Compliant Boundaries," MIT Report No. ASRL-TR-116-1 (1964)] analyzed the stability characteristics of the problem by solving for the eigenvalues of the problem. The numerical solution of the eigenfunctions of the Orr–Sommerfeld equation has always been a very difficult and tricky problem. Davey ["An Automatic Orthonormalization Method for Solving Stiff Boundary Valve Problems," J. Comput. Phys. **51**, 343–356 (1983)] has developed a technique which allows for quick solution for the eigenvalues *and the eigenfunction* of the problem at any Reynolds number. This technique has been applied to the problem at hand and it has given solutions in a quick and efficient manner. The results for the stability of the boundary layer disturbances are presented in the form of velocity profiles (eigenfunctions) within the fluid and the solid layer.

11:10

G9. Plumbing noise measurement and isolation. Jerry P. Christoff (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

Plumbing noise measurements on valves used for domestic lavatories, showers, tubs, and toilets have been measured utilizing the approach outlined in ISO Standard 3822. This technique evaluates the flow induced structural vibration transmitted into the piping system. The results clearly indicate that the principal source of noise is turbulence within the valve mechanism and not due to excessive velocities in the piping. Comparisons will be provided between American and European products. Several standard methods of isolating pipe wall vibration from building components are also evaluated.

11:25

G10. Hydroelastic analysis of multiple circular cylinders in an inviscid incompressible flow. W. H. Lin (AT&T Bell Laboratories,[a] Indian Hill, IL 60566)

This paper presents a theoretical analysis and numerical results of hydroelastic oscillations of a group of parallel, circular, elastic cylinders in an inviscid incompressible flow. On the assumption that the turbulent fluid loading is negligible, the principal excitations considered are the steady hydrodynamic force due to the existence of the cylinders in the smooth flow and the self-excited hydrodynamic forces caused by the motion of the cylinders. These hydrodynamic forces are obtained by solving a Lapace's equation for a potential function with Neumann conditions on the cylinders and the finiteness condition at infinity. The cylinders are assumed to be slender-beam rods with straight axes so that the hydrodynamic forces acting on the axes do not produce any twisting and the equations of motion of the cylinders' flexure can be approximated by slender-beam theory. The hydroelastic equations describing the interaction between the fluid motion and the cylinders' motion are casted into a matrix form and solved by the method of matrix inversion with the aid of digit computers. The critical flow speeds are determined by solving a complex eigenvalue problem associated with the coefficient determinant of the matrix equation with the method of iteration. Numerical results show that the added mass coefficients and the hydroelastic coefficients are symmetrical, and that the critical flow speeds depend much on the number, configuration, and orientation of the cylinders. [a] This work was performed during the author's tenure with Argonne National Laboratory, Argonne, IL 60439.

11:40

G11. Reducing the flow-induced self-noise of a sonar: A successful enhancement of experimental and numerical models. Bernard Garnier and Jacqueline Larcher (Métravib R.D.S., B.P. 182, 69132 Ecully Cédex, France)

The noise created by an external turbulent boundary layer in the heavy fluid contained in a sonar dome strongly depends on the dynamic properties of this shell. This work was performed to define the best design of the shell for a given shape, a given flow velocity, and a given sonar frequency range to minimize the flow-induced self-noise blinding the sonar in passive listening operational situations. Prior steps of this work were presented to the 106th Meeting of ASA [J. Larcher, J. Acoust. Soc. Am. Suppl. 1 74, S48, S78 (1983)] but they only reported simulations results. Since that time, operational prototypes have been measured out at sea; the results observed on operative sonars corroborate the whole predictive tool, which efficiently imbricates simplistic theoretical models with experimental simulations of the hydrodynamic excitation on 1/4 scaled or 1/1 sonar dome prototypes.

11:55

G12. Flow-induced noise and vibration of confined jets. Kam W. Ng (Launcher and Missile Systems Department, Naval Underwater Systems Center, Newport, RI 02841)

An experimental program was conducted in the acoustical water tunnel to determine the flow-induced noise and vibration on pipes due to various flow restrictors. Several flow restrictor configurations were tested for a range of flow velocities (up to 15 m/s). The flow configurations tested include: pipe flow, and flow restrictors with circular, coannular, rectangular cross-sectional areas, as well as multiple circular and slot jets. Wall pressure fluctuation and acceleration measurements were made using miniature hydrophones, flush-mounted on the inside wall of the pipe, and accelerometers mounted on the outside wall. Spectral and cross-spectral densities of the wall pressure and acceleration signals were determined. Experimental results showed that the flow-induced noise levels vary with the pipe axial location. The peak noise is located at the vicinity of the end of the jet potential core (six jet diameters downstream of the jet). Correlation of noise versus velocity showed a velocity to the 4.8th power relationship. Normalized noise spectra were obtained for the various flow configurations. The spectral shapes of the various flow configurations are quite similar, except that the coannular and slots jets show more high-frequency noise. The pipe-wall structural resonances were identified by correlating the various hydrophone signals. These resonance frequencies were consistent with the results obtained by impact testing. Furthermore, results from the correlation of hydrophone signals showed the existence of coherent structures, which probably control the generation of turbulence generated noise, near the exit of the jet.

Session H. Speech Communication I: Automatic Speech Recognition; Speech Processing

Jeffrey L. Elman, Chairman
Department of Linguistics, University of California, La Jolla, California 92093

Contributed Papers

8:30

H1. Current military/government applications for speech recognition. James W. Hicks, Jr. (SCI Technology, Inc., 8600 S. Memorial Parkway, Huntsville, AL 35802)

This paper presents an overview of several military/government programs in which SCI Technology, Inc. has implemented and tested its speech recognition system. Included are: (1) the Speckled Trout (U.S. Air Force), (2) LHX (Light Helicopter Experimental, U.S. Army), (3) Space Shuttle (NASA), (4) Space Station, (5) AFTI F-16, and (6) Advanced Tactical Fighter (ATF). Some programs consist of technology demonstrations, while others involve flight testing, and one, Speckled Trout, operationally installing and utilizing a system on a continual basis. In some cases, the hardware consists of a SCI Voice Control Unit (VCU-5137) and others, a Voice Development System (VDS-7001). For example, the Space Station application consisted of implementing a VDS in a power system monitoring and switching network in which power switches and loads could be controlled by verbal commands. In another application, two VCUs have been delivered to NASA for future Shuttle flights in which the remote controlled cameras on board will be manipulated (switched, positioned, and focused) using speech, freeing the astronauts' hands for more manually oriented tasks such as controlling the mechanical arm in the cargo bay. The LHX applications included interrogating various on-board systems (e.g., electrical, hydraulic, transmission) by speech commands and having the system status appear on video monitors. The Speckled Trout program is the first operational speech recognition system to be installed in an aircraft as an integral component of the aircraft's systems. Three VCUs are part of an integrated radio and navigational aides control system in which the system control can be either manual or by voice commands. However, since installation several months ago, the crew reports that radio control has been carried out almost exclusively by verbal commands. The AFTI F-16 program includes integrating a VDS as part of a flight simulator for testing and evaluation. The ATF program has only recently started and will involve integration as part of an advanced cockpit. This paper will also discuss the evolutionary process that has proven essential to the successful application of speech recognition technology into military and governmental systems of the future.

8:42

H2. Decoding the speech code—Applications of temporal decomposition. Stephen M. Marcus (AT & T Bell Laboratories, Murray Hill, NJ 07974 and Institute for Perception Research—IPO, Eindhoven, The Netherlands) and Bishnu S. Atal (AT & T Bell Laboratories, Murray Hill, NJ 07974)

Articulatory phonetics describes speech as a sequence of overlapping articulatory gestures, each of which may be associated with a characteristic ideal target spectrum. In normal speech, the idealized target gestures for each speech sound are often never attained, and the speech signal exhibits only transitions between such (implicit) targets. It has been suggested that the underlying speech sounds can only be recovered by reference to detailed knowledge of the gestures by which individual speech sounds are produced. It will be shown that it is possible to decompose the speech signal into overlapping "temporal transition functions" using techniques which make no assumptions about the phonetic structure of the signal or the articulatory constraints used in speech production. Previous work has shown that these techniques can produce a large reduction in the information rate needed to represent the spectral information in speech signals [B.S. Atal, Proc. ICASSP 83, 2.6, 81–84 (1983)]. It will be shown that these methods are able to derive speech components of low bandwiths that vary on a time scale closely related to traditional phonetic events. Implications for perception and the application of such techniques both for speech coding and as a possible front end for speech recognition will be discussed.

8:54

H3. Discovering acoustic features with a connectionist learning algorithm. Jeffrey L. Elman and David Zipser (Department of Linguistics and Institute for Cognitive Science, University of California at San Diego, La Jolla, CA 92093)

A method for discovering acoustic features in speech using a parallel distributed processing (or "connectionist") learning algorithm is described. The approach involves representing speech patterns as distributed patterns of activation over a network of interconnected processing elements. The "generalized delta rule" (Rumelhart, Hinton, and Williams, 1986) is used to teach the model to autoassociate various speech patterns with themselves. Teaching is accomplished by adjusting connection strengths between processing elements in order to minimize the error in autoassociations. This method was used to discover a set of acoustical features which efficiently and accurately encode speech sounds. The results of several simulations are reported, and the implications of the approach for models of human speech perception, as well as possible applications for machine-based speech recognition systems are discussed. [Work supported by the Office of Naval Research.]

9:06

H4. How practice affects speaker consistency in an automatic speech recognition setting. Linda A. Roberts, Dennis E. Egan, Jay G. Wilpon, and Jean Bakk (AT & T Bell Laboratories, Murray Hill, NJ 07974)

Our primary goal was to assess whether practice would enhance speaker consistency in an automatic speech recognition (ASR) setting. A second goal was to observe whether specific directions for successful ASR interaction [see L. A. Roberts, J. G. Wilpon, D. E. Egan, and J. Bakk, J. Acoust. Soc. Am. Suppl. 1 77, S89 (1985)] would improve speaker consistency throughout extended practice. Inexperienced ASR users spoke a single list of isolated words in ten daily sessions. The main result of practice was that subjects learned to form templates efficiently. No improvement over time was found for recognition accuracies or distances between utterances and their templates, likely because no feedback was provided. When utterances were compared to first-day templates, distances scores gradually increased up to the fourth testing session, after which time no further degradation was observed. Speakers who received directions were more consistent than those who did not, throughout all practice sessions. This result suggests that the directions produced behavior changes that could not be induced by practice alone.

9:18

H5. Lexical stress and speech recognition. Roberto Pieraccini (CSELT—Centro Studi e Laboratori Telecomunicazioni S.p.A. Via G. Reiss Romoli 274–10148 Torino, Italy)

The characteristics of lexical stress and their possible use for automatic speech recognition are analyzed in this paper, with reference to the Italian spoken language. A theoretical analysis of the constraints imposed by the stress in a large lexicon access strategy was conducted on the basis of a 12 000 word vocabulary. Additionally, an investigation on several thousands of words spoken by different speakers was made in order to extract the statistical properties of the main stress correlates. An algorithm based on a statistical model succeeds to detect with a good reliability the stressed vowel of an isolated utterance. In the continuous speech case the same model can be used to compute a measure of the likelihood that a vowel is stressed. In the first case, the hypothesized stress position could be used as a source of constraints in a large lexicon access problem. In the second case, efficient pruning strategies based on highly reliable stress decisions can be designed for continuous speech understanding systems.

9:30

H6. Effect of the spectral model order in automatic speech recognition. Kazuhiro Tsuga and Hynek Hermansky (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

It has been observed that in speaker-independent multi-template digit recognition, the 5th-order perceptually based LP (PLP) analysis method yields about 40% lower error rates than does the standard 14th-order LP. In speaker-dependent recognition, the 5th-order PLP model yields essentially the same recognition accuracy as do high-order PLP or standard LP models. In order to clarify the reasons for this result, RPS distances for LP and PLP models of single-frame phoneme-like spectra of male and female speakers were computed and arranged into distance matrices. Single-speaker 10th-order model matrices were adopted as the reference matrices for both LP and PLP methods. Cross-speaker PLP matrices become more similar to PLP reference matrices as the model order of cross-speaker matrices decreases to the 4th order. LP cross-speaker matrices remain relatively similar to the 5th order. Single-speaker PLP matrices remain relatively similar to the 5th order of the model while the single-speaker LP matrices of lower order differ significantly from the high-order ones. These results agree well with recognition rates obtained in the cross-speaker alpha-digit recognition for the same speakers. Results support the F1, F2' theory of speech perception.

9:42

H7. Use of articulatory signals in automatic speech recognition. L. D. Braida (IBM Thomas J. Watson Research Center, Yorktown Heights, NY 10598 and Research Laboratory of Electronics, MIT, Cambridge, MA 02139), M. A. Picheny, J. R. Cohen (IBM Thomas J. Watson Research Center, Yorktown Heights, NY 10598), W. M. Rabinowitz, and J. S. Perkell (Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

Automatic speech recognition systems generally attempt to determine the spoken message from analysis of the acoustic speech waveform. In this research we evaluated the performance of the IBM Speech Recognition System [F. Jelinek, Proc. IEEE 73, 1616–1624 (1986)] when the input included measurements of selected articulatory actions occurring during speech production. The system achieved significant recognition rates (for isolated words in sentences) when the acoustical signal was disabled and the input was restricted to articulatory signals similar to those sensed by users of the Tadoma method of tactile-speech recognition [e.g., Reed *et al.*, J. Acoust. Soc. Am. 77, 247–257 (1985)]. In other tests the availability of articulatory inputs improved recognition performance when the acoustical signal was sufficiently degraded by additive white noise. Improvements were observed independent of whether the recognition system made use of the likelihood that words would appear in the vicinity of the other words in a given sentence.

9:54

H8. Grammatical constraints and recognition performance. P. J. Price, Y.-L. Chow, M. O. Dunham, O. Kimball, M. Krasner, F. Kubala, J. Makhoul, S. Roucos, and R. Schwartz (Speech Signal Processing Department, BBN Laboratories, Inc., 10 Moulton Street, Cambridge, MA 02238)

The integration of grammatical with acoustical knowledge sources in the BBN continuous speech recognition system, BYBLOS, and the resulting effects on performance are described. The system consists of feature extraction, acoustical scoring, and linguistic scoring. Feature extraction is based on vector quantized mel-warped cepstral coefficients. Acoustical scoring is derived from a hidden Markov model for each word, where word models are based on phonetic spellings so that models can be computed for words that have never been trained. The linguistic model is represented as a finite automaton derived automatically from a context-free specification of the task-domain syntax and semantics. It is shown how recognition performance varies with properties of the grammars. Word recognition accuracies of over 98% have been achieved in continuous speaker-dependent mode for 350-word tasks with grammars having maximum perplexity in the range of 20 to 60. [Work supported by DARPA and monitored by NAVELEX.]

10:06

H9. Evaluation of ASR front ends using synthetic vowel-like sounds. Hynek Hermansky and Hector Raul Javkin (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

Analysis techniques and distance metrics interact in their effectiveness as the front end of the automatic speech recognizer (ASR). Synthetic vowel-like sounds were used to compare different ASR front ends, focusing on cepstrally smoothed standard LP analysis and the recently proposed perceptually based LP (PLP) analysis combined with the LP cepstral distance and the root power sum (RPS) distance metrics. (1) The findings of Kamm and Kahn [J. Acoust. Am. Suppl. 1 78, S82 (1985)] that standard LP overemphasizes the importance of higher formats were confirmed. The tonality scale inherent in PLP representation alleviates this problem. (2) In PLP, as opposed to LP, the contribution of a formant change depends much more on the particular vowel in which it occurs. (3) LP is less sensitive to changes in formant bandwidth than PLP. (4) RPS is less sensitive to the overall spectral slope changes than the cepstral metrics. (5) With PLP, RPS emphasizes the dominant formant more than the cepstral metric. (6) All examined techniques are sensitive to the harmonic structure of speech. The re-analysis of published formant frequency jnd experiments suggests this sensitivity might not be inconsistent with human perception.

10:18–10:30

Break

10:30

H10. Spectral edge orientation as a discriminator of fricatives. Anthony Bladon and Franz Seitz (Phonetics Laboratory, University of Oxford, 37-41 Wellington Square, Oxford, OX1 2JF, United Kingdom)

Both "physical" and "auditorily transformed" spectra of English sibilant fricatives were investigated for evidence that might: (a) discriminate between /s/ and /ʃ/; (b) do so independently of vowel context; and (c) normalize the substantial speaker-sex differences found. Measurements included the frequency location of the first peak of fricative noise, the location of the low-frequency edge descending from this peak, the amplitude range of this descent, and its gradient or orientation. Analyses of variance on our data demonstrated a marked superiority for the orientation of the spectral edge, which differentiated the phonemes reliability, while remaining independent of vowel context or speaker sex. This was true, however, only of the edge as transformed in auditory space. Given the analogous ability of the eye-to-track edges, the notation appears to have some plausibility for speech perception theory and an obvious implication for automatic speech recognition. Perceptual confirmation of the importance of spectral edges in fricatives is now being sought.

10:42

H11. Toward automatic recognition of the semivowels /r, w, l, y/: A progress report. Roy W. Gengel and James L. Hieronymus (Institute of Computer Science, Building 225, National Bureau of Standards, Gaithersburg, MD 20899)

The goal is to develop a speaker-independent, semivowel detector–classifier for use with continuous speech. Preliminary analysis of semivowels within continuous speech indicates large within-(and between) speaker variability in the respective semivowel formant frequencies, amplitudes, and durations of steady states and transitions. Deviations from textbook descriptions based on isolated words are significant. Coarticulation effects are also large. Nevertheless, through the use of energy, energy ratios, zero crossings, and signal envelopes, a logical analytical scheme was developed that classifies segments of continuous speech as: semivowel, high-front vowel, vowel, nasal, fricative, stop, or plosive. The specific characteristics of this scheme will be discussed and the computer-generated displays that were used will be presented. Results of using this scheme on a large corpus of phonetically labeled speech will be presented. [Work supported, in part, by DARPA.]

10:54

H12. Location and classification of plosives, affricates, and fricatives in natural continuous speech. Robert A. Brennan, Benjamin Chigier, and Ronald A. Cole (Department of Computer Science, Carnegie Mellon University, Pittsburgh, PA 15213)

Algorithms have been developed to locate and classify plosive ([n], [d], [g], [p], [t], and [k]), affricate ([ch] and [jh]), and fricative ([f], [th], [s], [sh], and [z]) segments in natural continuous speech. The location algorithms use rules to locate target segments. The classification algorithms use feature measurements values and multivariate classifiers to assign a label probability to each hypothesized segment. The algorithms were tested on 400 utterances produced by 40 speakers that were not used to train the system. Performance of the location and classification algorithms was compared to human performance; i.e., segment boundaries and labels provided by experienced labelers. Location results will be reported in terms of the percentage of target segments located, the number of additional segments hypothesized, and the average left and right boundary error. Classification results will be presented in terms of rank order statistics, the average probability assigned to each segment, and confusion matrices. [Work supported by NSF and DARPA.]

11:06

H13. Glottal inverse filtering of nasalized vowels. Paul Milenkovic and Feng Mo (Department of Electrical and Computer Engineering, University of Wisconsin—Madison, 1415 Johnson Drive, Madison, WI 53706)

Linear prediction provides a means for automatically determining an inverse filter for estimating the glottal volume velocity waveform from the acoustic speech waveform. The behavior of this type of glottographic analysis is considered for nasalized vowels. The analysis is applied to non-nasal and nasal productions that are synthesized using a transmission line model of the vocal and nasal tracts. The analysis is also applied to nasal and non-nasal vowel productions made by a male speaker. The effect of the nasalization is to introduce extraneous ripple onto the closed phase portion of the glottal volume velocity waveform without changing the observed open quotient. This ripple results from a closely spaced resonance antiresonance pair that is not compensated by the all-zero linear prediction inverse filter. The low level of the ripple is accounted for by the narrow frequency range over which the inverse filter is in error. [Work supported by NIH.]

11:18

H14. Vocal tract areas from LPC—A comparison of several techniques. J. N. Larar, J. Schroeter, and M. M. Sondhi (AT & T Bell Laboratories, Murray Hill, NJ 07974)

The need for good initial estimates of cross-sectional vocal tract areas [e.g., Schroeter, Larar, and Sondhi, J. Acoust. Soc. Am. Suppl. 1 **80**, S19 (1986)] has renewed our interest in LPC techniques. Starting with speech synthesized from a vocal tract model with known cross-sectional areas [Schroeter and Sondhi, J. Acoust. Soc. Am. Suppl. 1 **78**, S6 (1985)], LPC-derived areas that closely approximate the areas used to control the synthesizer are sought. In a comparative study,the LPC method used to derive the predictor polynomial (autocorrelation, covariance, and stabilized covariance methods) as well as its particular application with respect to location of pitch epochs is varied; pitch synchronous and asynchronous analyses are performed with different assumptions regarding the glottal termination. Spectral shaping and source components in the inverse filter are removed and formant shifts due to yielding wall losses are considered. Areas are computed using the standard Wakita recursion [Wakita, IEEE Trans. **ASSP-27**, 281–285 (1979)] proceeding from the glottis to the lips with the first supraglottal section set to the known value (used for synthesis). Results are presented for steady vowels and for voiced segments produced with a time varying vocal tract.

11:30

H15. A formant tracker based on pitch synchronous spectra and bark scaling. James L. Hieronymus and William J. Majurski (Institute for Computer Sciences and Technology, Building 225, Room A216, National Bureau of Standards, Gaithersburg, MD 20899)

A formant tracker for continuous speech recognition which uses pitch synchronous spectra has been developed. The spectra are bark scaled to control diffuseness in the higher formants. Then the longest and strongest contiguous ridges are found beginning with regions of high intensity (vocalic nuclei). These are labeled as possible formant trajectories. In regions where the ridges are discontinuous, an attempt is made to join ridges of approximately the same amplitude and frequency. Then the formants are assigned, based on the average frequencies of the ridges in the vocalic regions. Statistically based heuristics are used when several candidate ridges occupy the region where formant frequencies overlap. Difficult areas include nasalized vowels and fronted back vowels. Results of evaluating the formant tracker on a large number of continuous utterances will be presented. [Work supported, in part, by DARPA.]

11:42

H16. Speech analysis and synthesis using a vocal tract/cord model. Juergen Schroeter, Jerry Larar, and Man Mohan Sondhi (AT & T Bell Laboratories, Murray Hill, NJ 07974)

The use of our vocal cord and tract model [Schroeter and Sondhi, J. Acoust. Soc. Am. Suppl. 1 **78**, S6 (1985)] for speech coding at bit rates below 4.8 kb/s is being considered. Earlier work [Flanagan *et al.*, J. Acoust. Soc. Am. **68**, 780–791 (1980)] reported on applying an unconstrained optimization method for jointly finding the best vocal tract and glottal parameters directly from the speech signal (one-stage optimization). This method has been extended by optimizing tract and glottal parameters sequentially (two-stage optimization). In a first stage, a search was made for the parameters of an inverse (tract) filter that give the best match between the inverse filtered speech and the output of the vocal cord model for a given set of glottal parameters. In a second stage, the parameters of the cord model were optimized for a given set of tract parameters so that the synthesized speech matches the original speech as closely as possible. This method has the advantage of separating the source and tract into different optimization stages. By doing so, the inherent constraints of the respective models can be easily used to constrain the search space. Compared to the one-stage optimization, the search required in each stage is over a parameter space of a smaller dimension. Different start-up procedures for the two-stage optimization are investigated. Results are compared to those of the one-stage optimization.

11:54

H17. Coherence-based enhancement of noisy speech. Les E. Atlas and Alexandra J. Clayton (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

A standard coherence measure has been used to give an estimate of signal-to-noise ratio (certainty indicators) as a function of frequency. The two spectra which are used for this measure are the noisy speech spectrum and a model spectrum derived from pitch period estimates of the noisy speech. It is intended that the certainty indicators be used as either weights for matches in automatic recognition or as multiplicative weights for reconstruction of speech with improved intelligibility. Two cases of noisy speech have been studied: single words contaminated by bandpass-filtered white noise (S1) and sentences contaminated by unfiltered white noise (S2). For case S1 it has been found that the certainty indicators will pinpoint the noisy band for voiced speech and will, as expected, not reliably indicate the noisy band for unvoiced speech. For case S2, preliminary results indicate that the quality of the enhanced speech has dropped, yet the intelligibility is slightly improved. [Work supported by NSF and Boeing Computer Services.]

12:06

H18. Adaptive filtering for speech enhancement. P. M. Clarkson, P. R. White, and J. A. Mardell (Institute of Sound and Vibration Research, University of Southampton, Southampton SO9 5NH, United Kingdom)

Several authors have considered the use of adaptive noise cancellation for speech enhancement. These efforts have concentrated on the simple noisy gradient descent (LMS) algorithm and, in the absence of an external reference measurement, have formed a reference internally using a delayed version of the input. Limited improvements in SNR (for voiced speech enhancement) have resulted due to the inability of the filter to cope effectively with variations in power (of the speech) across frequency, discrete changes in the pitch period and because of the limited SNR of the reference. This study has approached the problem by replacing the noisy reference measurement by a synthetic reference which consists of a low-pass filtered series of discrete impulses at intervals corresponding to the pitch period of the input. This method has been compared with a modified version of the usual noise canceller (using an exact least-squares algorithm) on a sustained vowel segment. Higher SNR improvements for the synthetic reference have been measured. However, the method does require an estimate of the pitch period and appears quite sensitive to errors in this estimate.

12:18

H19. Intelligibility of ICAO spelling alphabet words as a function of LPC narrow-band processing and bit error rate. A. Schmidt-Nielsen (Naval Research Laboratory, Code 7520, Washington, DC 20375)

The intelligibility of ICAO spelling alphabet words was tested to determine the kinds of confusions that occur under severely degraded digital voice conditions. The test conditions included a 2400 bit/s linear predictive coding (LPC) voice processing algorithm with random bit error rates of 0%, 2%, 5%, 8%, and 12%. The ICAO alphabet (ALFA, BRAVO, CHARLIE,...) was developed for communication under degraded conditions, and the words have been selected with the object of minimizing confusions. Scores on a standard intelligibility test, the Diagnostic Rhyme Test (DRT), were obtained for comparison and to determine a prediction function from DRT scores to spelling alphabet performance. The types of confusions that were obtained with the narrow-band digital speech with bit errors showed a number of substantial differences from the types of errors normally found with spelling alphabet words tested in noise. The most frequent error was that PAPA was heard as ALPHA (but ALPHA was virtually never heard as PAPA). As is to be expected, scores on the spelling alphabet words fell monotonically with falling DRT scores but were considerably higher than the DRT scores for the same conditions up to 5% errors; after that, the spelling alphabet scores dropped off rapidly to levels as low as the DRT scores.

TUESDAY MORNING, 9 DECEMBER 1986 — SALONS G AND H, 9:00 TO 11:15 A.M.

Session I. Underwater Acoustics I: Quality Assessment of Numerical Codes, Part 1: Codes

Leopold B. Felsen, Chairman

Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, New York 11735

Chairman's Introduction—9:00

Invited Papers

9:05

I1. The art of generating meaningful results with numerical codes. Finn B. Jensen (SACLANT ASW Research Centre, 19026 La Spezia, Italy)

Today numerical models are widely used within the underwater acoustics community for solving complex wave-propagation problems. While it is easy for anybody to produce "interesting" field solutions with a model, it is extremely difficult even for experts to generate numerical results that can be considered an accurate solution to a stated acoustical problem. This difficulty derives from insufficient knowledge both of the approximations introduced to formulate a solvable set of equations and of the accuracy and convergence problems associated with the numerical implementation itself. When experimental data are not available for checking a numerical solution, there are, in principle, only two ways to gain confidence in the numerical result (it is here assumed that straightforward checks of reciprocity and energy conservation have been performed): (1) Use a different model to confirm the validity of the original solution; or (2) compare the numerical result to an accepted reference solution to a similar propagation problem. Currently, the intermodel comparison is the only viable approach, since few reference solutions are available, none of which is for general range-dependent problems. Illustrative examples of both easily generated incorrect field solutions and artfully constructed correct field solutions will be given for range-independent environments using computer codes based on normal-mode and fast-field theory, and for range-dependent environments using codes based on adiabatic modes, coupled modes, and the parabolic equation.

Contributed Papers

9:30

I2. An identification of the approximations in stepwise coupled modes. Richard B. Evans (ODSI Defense Systems, Inc., North Stonington Professional Center, Route 2 and 184, North Stonington, CT 06359)

The stepwise couple mode method was developed to compute the acoustic field in a range-dependent ocean environment. The goal of this development was to eliminate the one-way and limited aperture assumptions that are made in the parabolic approximation. Having eliminated these approximations, it was expected that the stepwise coupled mode solution could serve as benchmark for range-dependent propagation in the same way that the normal mode solution serves as benchmark for range-independent propagation. Any numerical solution has some inherent approximations and associated conditions under which the resulting errors are small. The approximations and errors should be well understood in a reliable benchmark. The purpose of this talk is to begin to address this issue by identifying the approximations in the stepwise coupled mode method. Some of the errors which result form these approximations will also be discussed. [Work supported by NORDA.]

9:45

I3. A detailed examination of the path integral-based phase-space-marching algorithm for wide-angle, one-way wave propagation. Louis Fishman (Department of Civil Engineering, The Catholic University of America, Washington, DC 20064) and Stephen C. Wales (Acoustics Division, Naval Research Laboratory, Washington, DC 20375)

The analytical approximations and numerical implementation of the path integral-based phase-space-marching algorithm are examined in detail. The approximate Weyl symbol constructions are evaluated with respect to the variations of the refractive index field on the wavelength scale, with attention focused on the particularly difficult case of very strong gradients. The local error, energy flux conservation, and effective stability of the numerical algorithm are discussed. Phase-space-filtering strategies are analyzed and illustrated with numerical computations. Detailed considerations involving the startup field, alternating false bottom, and range and depth step-size dependence are presented.

10:00

I4. Benchmark problems used to test the implicit finite difference (IFD) model. Ding Lee, George Botseas (Naval Underwater Systems Center, New London, CT 06320), and Donald F. St. Mary (Center for Applied Mathematics and Mathematical Computation, Department of Mathematics and Statistics, University of Massachusetts, Amherst, MA 01003)

The implicit finite difference (IFD) model was developed approximately 4 years ago to predict ocean acoustic wave propagation under variable environments. Its primary objective was to be "accurate" and "general purpose." Since then, the model's capability has been improved. It has been widely used for research as well as for applications. During the test state for validity, a number of known solutions were used for reference comparisons. These test problems can be used as a severe test of a model's ability to handle variable environments accurately. This paper exhibits two test problems: One tests accuracy; the other demonstrates capability. In order to assess the computer code, how the mathematical model was developed, and how the method of finite differences was introduced along with an analysis of its accuracy, stability, and convergence are reviewed. Then, how the numerical solution was implemented into computer codes is discussed. Finally, the question is asked: What can this model do for you?

10:15

I5. Critical assessment of a marching method for an elliptic envelope equation in underwater sound propagation. George Knightly and Donald F. St. Mary (Center for Applied Mathematics and Mathematical Computation, Department of Mathematics and Statistics, University of Massachusetts, Amherst, MA 01003)

A scheme for computing propagation loss in the farfield of a rectangular ocean waveguide (soft or hard bottom) by marching in range using the farfield elliptic envelope equation is assessed. The method has shown some promise, despite the ill-posedness of the underlying initial value problem. A theoretical analysis of the problem is presented indicating parameter ranges where the method works. Approximate solutions obtained by the method are compared with those obtained by other methods.

10:30

I6. Benchmark models by the method of finite differences for VLF pulse propagation in bottom-interacting ocean acoustics. R. A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A suite of benchmark models are proposed for VLF pulse propagation in bottom-interacting ocean models. The models are: (i) a homogeneous medium; (ii) a flat seafloor with a sharp transition from water to basalt; (iii) a flat seafloor with a linear gradient in elastic parameters between water and basalt; (iv) a flat seafloor with a linear gradient in elastic parameters overlying a sharp transition to basalt; (v) a flat seafloor with a uniform sediment layer overlying a basaltic basement with continuously increasing elastic parameters with depth; (vi) a range-dependent model of water over Plexiglas with a step in the Plexiglas offset from the source; and (vii) a range-dependent waveguide problem for a shot over the continental shelf propagating out over the continental slope. Solutions to all of the above problems have been obtained using finite difference solutions to the elastic wave equation. The range-independent models have been solved using discrete wavenumber techniques. All models are based on physically realistic models for which either laboratory or field data are available.

10:45

I7. Three-dimensional finite element analysis of waves in an acoustic medium with inclusion. Yu-chiung Teng (Aldridge Laboratory of Applied Geophysics, Columbia University, New York, NY 10027)

By using the pressure field as the basic unknown variable, two three-dimensional acoustic finite element algorithms computer codes, viz., ACOU3.A and ACOU3.B, have been implemented. Code ACOU3.A is designed for the uniform-grid space discretization such that the storage of the global pseudostiffness matrix is reduced to a single column vector of "27" elements, applicable to any inhomogeneous media. Likewise code ACOU3.B is designed for the irregular-grid space descretization and possesses the same feature of a reduced global pseudostiffness matrix of ACOU3.A. By using ACOU3.A, three $30\times30\times30$ elements models were investigated, including (i) a half-space, (ii) a half-space with a $10\times7\times5$ elements inclusion, and (iii) a half-space with a $10\times2\times5$ elements inclusion, where the inclusion has the physical property different from the half-space. In order to demonstrate the third dimension effects on wave propagation, the above posed three-dimensional problem to the equivalent axial-symmetric and two-dimensional approximation models was reduced. By using a full three-dimensional algorithm, the amplitude of the reflected waves carries the information of the third dimension thickness of the inclusion, which cannot be obtained by either the axial-symmetric, or the two-dimensional approximations.

11:00

I8. Convergence rate of codes for numerical quadrature techniques for classical ray tracing. Edward R. Floyd (Arctic Submarine Laboratory, Naval Ocean Systems Center, San Diego, CA 92152-5000)

The estimated residual error (or its bound) for numerical quadratures is usually expressed in terms of a derivative of some order of the integrand or some residual factor of the integrand after factoring out a countable number of zeros and singularities that occur along the integration path. The order of the derivative is a function of the number of sample points for evaluating the integrand. Regrettably, the magnitudes of these higher-order derivatives are difficult enough to estimate for even analytic sound velocity profiles. In practice, observed sound velocity profiles, which are usually given in tabular form and include measurement errors, exacerbate our inability to assess the magnitudes of these higher-order derivatives. An estimate of the residual error expressed in terms of a first derivative would be far more practical for both analytic and observed sound velocity profiles.

Session J. Architectural Acoustics II and Musical Acoustics II: Acoustical Evaluation of Halls for the Performing Arts, Part 2

Carleen M. Hutchins, Chairman
Catgut Acoustical Society, 112 Essex Avenue, Montclair, New Jersey 07042

Chairman's Introduction—2:00

Invited Papers

2:05

J1. User evaluations of the acoustical characteristics of recently completed performing arts halls. Donald L. Engle (Engle Arts Consultants, Inc., 2913 Eagle Estates Circle South, Clearwater, FL 33519)

The evaluations of individuals who respond to, and live with, the acoustics of recently completed performance arts spaces have been surveyed. The views surveyed are those of performers, media critics, reporters, facility managers, and patrons. The ways in which such evaluations are used in selecting an acoustical consultant for a new or renovated facility are discussed.

2:35

J2. The neglected role of musicians in the preliminary planning of concert halls. Robert Finn (Music Critics Association and Music Critic for the Cleveland Plain Dealer, 1801 Superior Avenue, Cleveland, OH 44144)

A common complaint among musicians is that they are almost never involved in the preliminary planning stage when a new concert hall is being built. The basic planning which determines the acoustical properties of the finished building is thus left to architects, acousticians, and engineers, who are usually nonmusicians. There are certain criteria which the musicians—who must, after all, live with the hall long after the architects and acousticians have departed—consider basic. There are building materials they prefer for acoustical reasons, and configurations they consider better than others. What are these considerations, and how subjective are they? Do they have any validity when measured against the scientific data of the experts? Is it possible for the scientific community to learn from the experiences of practicing musicians who play in the halls they design—and conversely, would the musicians profit by paying closer attention to the work of scientists and acousticians?

3:05–4:05

Panel Discussion

Panel Members

Martin Bernheimer
Music Critic, Los Angeles Times

Thomas O'Connor
Architecture Critic, Orange County Register

Theodore J. Schultz
Acoustical Consultant, Boston, Massachusetts

Session K. Biological Response to Vibration II and Physical Acoustics II: Physical Mechanisms of Biological Effects of Sound and Vibration, Part 2

Floyd Dunn, Chairman
Bioacoustic Research Laboratory, 1406 West Green Street, University of Illinois, Urbana, Illinois 61801

Contributed Papers

2:00

K1. Acoustic cavitation generated by an extracorporeal shockwave lithotripter. Lawrence A. Crum,[a)] Mary Dyson (Division of Anatomy, United Medical and Dental Schools, Guy's Campus, London Bridge, London SE1 9RT, England), Andrew J. Coleman, and John E. Saunders (Department of Medical Physics, St. Thomas's Hospital, Lambeth Palace Road, London SE1 7EH, England)

Evidence is presented of acoustic cavitation generated by a Dornier extracorporeal shockwave lithotripter. Using x-ray film, thin aluminum sheets, and relatively thick metal plates as targets, evidence of liquid jet impacts associated with cavitation bubble collapse was observed. The jet impact was violent enough to puncture thin foils and deform metal plates. Furthermore, numerous jet impacts were generated over a volume greater than 200 cm^3. It is likely that such violent cavitation will also occur in tissue, and observed biological effects, e.g., renal calculus disintegration and tissue trauma, may be related to cavitation damage. [Work supported in part by the ONR, NIH, NSF, and the Fulbright Commission.] [a)] Permanent address: Department of Physics, Accelerator Building, University of Mississippi, University, MS 38677.

2:15

K2. Sonoluminescence produced by steady cavitation. D. F. Gaitan (Physical Acoustics Research Group, Department of Physics and Astronomy, University of Mississippi, Oxford, MS 38677)

Sonoluminescence (SL) is generally attributed to the radiative recombination of hydroxyl free radicals produced by the high temperatures and pressures associated with cavitation bubble collapse. Therapeutic ultrasound systems are known to produce large numbers of free radicals in water and biological fluids and thus present a possible health risk. In an examination of light produced by acoustic standing wave configurations at various frequencies ranging from 20 kHz–1 MHz, steady light emissions have been observed from the cavitation field indicative of violent bubble pulsation rather than cavity collapse. In many cases, multiple flashes occur each cycle, always maintaining a fixed phase with respect to the driving acoustic pressure. This "steady" cavitation is attributed to violent bubble pulsation, and an explanation of this phenomenon will be attempted in terms of numerical studies of bubble dynamics. [Work supported in part by the ONR and the NIH.]

2:30

K3. Transient pulsations of cavitation bubbles. H. G. Flynn (Department of Electrical Engineering, University of Rochester, Rochester, NY 14627) and Charles C. Church (Department of Biophysics, University of Rochester, Rochester, NY 14627)

Transient behavior of small gas bubbles in a liquid set into violent motion by ultrasonic pressure waves is of interest because of widespread use of microsecond pulses in diagnostic ultrasound. Such pulses contain only a few pressure cycles and the transient pulsations of bubbles set in motion by such pulses would determine the bubble–ultrasound interaction. A computer study has been made to obtain a global representation of the pulsation amplitudes $R(t)$ of small gas bubbles (nulcei) in water during the first few cycles of a cw ultrasonic pressure. One objective was to obtain a better understanding of cavitation phenomena where many nuclei with initial radii from 0.1 to 20 μm are set in motion at pressures ranging from 0.5–5 bars and at frequencies from 0.5–10 MHz. Results allowed construction of surfaces showing the relative bubble amplitude R/R_n as a function of R_n and of the time t/T_A, where T_A is the acoustic period. One finding is that, in the range of peak pressures found in diagnostic pulses, transient cavities would be generated during the first pressure cycle from nuclei with initial radii as small as a few microns (μm). [Research supported by NIH.]

2:45

K4. Direct observation of the forced radial oscillations of single cavitation bubbles. R. G. Holt (Physical Acoustics Research Laboratory, Department of Physics, University of Mississippi, Oxford, MS 38677)

The radial response of a bubble in a liquid subject to an acoustic field has long been the object of considerable interest [see, for example, W. Lauterborn, J. Acoust. Soc. Am. **59**, 283–293 (1976); B. E. Noltingk and E. A. Neppiras, Proc. Phys. Soc. London Ser. B **63**, 674 (1950)]. However, the most often generated graph in these theoretical and numerical studies, the radius–time (R–T) curve, has never been experimentally observed until now. The first direct observation of the R–T response of a single bubble will be presented. The measurements were made on a bubble "levitated" [M. Strasberg, J. Acoust. Soc. Am. **33**, 359 (1961)] near the antinode of an acoustic stationary wave by digitizing the output of a photodiode monitoring the intensity of laser light scattered by the bubble. Comparisons will be made with the results of some recent numerical models. [Research supported by ONR.]

3:00

K5. The effect of nonlinear distortion of biomedical ultrasound on acoustic cavitation. Charles C. Church (Department of Biophysics, The University of Rochester, Rochester, NY 14642)

Previous theoretical studies of acoustic cavitation postulated as the driving pressure a Gaussian pulse, a sine wave within a Gaussian envelope, or a pure-tone sine wave. In this study, the harmonic components of a distorted sine wave are calculated following the method of Blackstock [J. Acoust. Soc. Am. **39**, 1019–1026 (1966)]. The magnitudes of these harmonics are attenuated as a function of frequency and distance from the transducer and are summed using a constant phase shift. This produces a waveform quite similar to those seen in the laboratory. Distorted waves produced at 1, 3, 5, and 10 MHz are used to derive Cramer's equations for nonlinear bubble dynamics [*Cavitation and Inhomogeneities in Underwater Acoustics*, edited by W. Lauterborn (Springer, New York, 1980), pp. 54–63]. Typical bubble radius versus time curves and transient cavitation thresholds ($R/R_n > 2.0$) are shown. Due to the shifting of energy from the fundamental to the harmonics and to attenuation, these thresholds are generally higher than for pure sine waves. [Work supported by NIH.]

3:15

K6. Thresholds for cavitation produced in water by pulsed ultrasound. Anthony A. Atchley,[a] Leon A. Frizzell,[b] and Robert E. Apfel (Department of Mechanical Engineering, Yale University, P. O. Box 2159 Yale Station, New Haven, CT 06520)

Initial results of an experimental investigation of transient cavitation produced by pulsed ultrasound are presented. Water is irradiated with a focused transducer and the subsequent cavitation detected acoustically by a second transducer. The detection technique relies upon the scattering of the irradiation field by the bubble cloud associated with the transient cavitation. A novel feature of the experimental apparatus is a system with which potential cavitation nuclei can be passed directly through the focal region of the irradiation transducer. The threshold for transient cavitation was measured for 1- and 2.25-MHz pulses having durations from a few cycles to 500 μs and repetition frequencies from 50 to 5000 Hz. The data are consistent with, but extend, those of Crum and Fowlkes [Nature **319**, 52–54 (1986)] in which sonoluminescence emission was used as the cavitation criterion. The results of the experiment are discussed with regards to the predictions of present theories, in particular, that of Apfel [IEEE UFFC 33, No. 2, 139–142 (1986)]. [Work supported by NIH. AAA acknowledges the support of the F. V. Hunt Postdoctoral Fellowship.] [a] Permanent address: Department of Physics Code 61 Ay, Naval Postgraduate School, Monterey, CA 93943. [b] Permanent address: Bioacoustics Research Laboratory, University of Illinois, 1406 W. Green Street, Urbana, IL 61801.

3:30

K7. Enhancement of protein synthesis by neuroblastoma cells exposed to ultrasound cavitation. Peter D. Edmonds, Pepi Ross, and Ruth M. Yamawaki (Bioengineering Research Laboratory, SRI International, Menlo Park, CA 94025)

Enhancement of the cavitational effect of iodine release from sodium iodide solution during repetitive 1-MHz tone burst excitation of the solution of a rotating test tube has been reported. [V. Ciaravino, H. G. Flynn, and M. W. Miller, Ultrasound Med. Biol. **7**, 159–166 (1981)]. This enhancement was attributed to concurrent operation of two mechanisms: depletion of small nuclei from the cw mode size distribution generated during a tone burst, and survival from the previous tone burst of small nuclei lying in the same size range that is depleted in the cw mode. Twenty-four hours after exposing C1300 neuroblatoma cells (N2A) in rotating tubes to 1-MHz ultrasound tone bursts [1:1, durations from 6 to 600 ms; 3.4 W/cm^2 spatial peak, burst average intensity, and 5 min total treatment duration (on + off periods)] at 37 °C, enhancement of protein synthesis compared to control cells was observed. Protein synthesis was measured by uptake of ^{3}H-leucine. The similarity between results observed for cavitation-stimulated iodine release and cellular protein synthesis is highly suggestive of cavitation as the cause for this biological effect. [Work supported by PHS.]

3:45–4:45

Bull Session

Open discussion of induced cavitation in water, gels, bio-suspensions, plants, insects, laboratory animals, and man.

TUESDAY AFTERNOON, 9 DECEMBER 1986 SALON 3, 1:30 TO 5:00 P.M.

Session L. Engineering Acoustics II: Transducers and Arrays

George S. K. Wong, Chairman

Physics Division, National Research Council, Montreal Road, Ottawa, Ontario K1A 0S1, Canada

Contributed Papers

1:30

L1. Experimental results of an underwater magnetohydrodynamic transmitting acoustic transducer. Stephen C. Schreppler and Ilene Busch-Vishniac (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

The magnetohydrodynamic transducer operates on the familiar principle of the Lorentz force which results from interaction of electromagnetic fields in a conducting medium. The transducer investigated has the Lorentz force generated within an acoustic waveguide by a uniform magnetic field and a mutually orthogonal time harmonic current density interacting in a sodium chloride–water medium. The waveguide is 10 cm in length with a uniform 3.8 cm$\times$3.2 cm rectangular cross section, and is excited in the plane-wave mode. Acoustic power produced in the waveguide is radiated from apertures at either end of the waveguide into a surrounding free medium producing a dipole radiation pattern. Measurements of the transmitting sensitivity and radiation patterns are in good agreement with theory [J. Acoust. Soc. Am. Suppl. 1 **78**, S52 (1985)]. [Research supported by the Office of Naval Research.]

1:45

L2. Resonant reciprocity calibration of an ultracompliant transducer. S. L. Garrett (Physics Department, Code 61 Gx, Naval Postgraduate School, Monterey, CA 93943) and G. W. Swift (Condensed Matter and Thermal Physics Group, P-10, Los Alamos National Laboratory, Los Alamos, NM 87545)

A reciprocity calibration technique for application to an ultracompliant acoustic transducer in a resonator will be presented. The technique was applied to a magnetohydrodynamic (MHD) transducer in an open (pressure release) trough filled with either salt water or mercury as the wave propagating medium. The agreement between the reciprocity calibration and a calibration based on a primary standard is excellent. This result demonstrates that the sensitivity of the MHD transducer is just that given by the simple application of Faraday's law, in contrast to recent measurements indicating otherwise [P. H. Moose and R. F. Klaus, J. Acoust. Soc. Am. **74**, 1066 (1983)]. This reciprocity technique should be applicable to calibration of a wide range of transducers, including hydrophones and accelerometers. [Work supported by the Office of Naval Research.]

2:00

L3. Design model of large, uniform, conformal arrays of bender bar and flextensional transducers. G. A. Brigham (Aquasonics, Inc., Anaheim, CA 92806)

The single element design technology of flextensional and bender bar transducers is several decades old but use of either type of transducer in conformal arrays presents newer and more formidable design problems. When the array is very large the effects of edge diffraction on the outer elements can be ignored to generate a zeroth ordered waveguide design model to estimate radiation loading at any steering angle. Highly eccentric shelled flextensionals and bender bar transducers are then included in one common format. Since both types are largely flexural, they are resonant at frequencies where the interelement spacing is much smaller than a wavelength in water and only the low-frequency inertial and plane-wave volume flow components of loading need be determined. Several array geometries have been studied and the reactive loadings calculated. This paper shows both the theoretical and numerical results of various mass loadings as functions of array and element geometry. [Research supported by the Naval Underwater Systems Center, New London; the Naval Ocean Systems Center, San Diego; the Electric Boat Division of General Dynamics.]

2:15

L4. Analysis of radiating flexural shell sonar transducers using the finite element method. B. Hamonic, J. C. Debus, J. N. Decarpigny (Institut Supérieur d'Electronique du Nord, 41 boulevard Vauban, 59046 Lille Cedex, France), D. Boucher, and B. Tocquet[a] (Groupe d'Etude et de Recherche de Détection Sous-Marine, Le Brusc, 83140 Six Fours les Plages, France)

New flexural shell transducers for low-frequency applications are currently developed which are characterized by a large volume velocity and a drastic reduction of their resonance frequencies as soon as they are flooded, due to added mass effect. To design these transducers, a finite element modeling is very useful, because it can accurately handle the assembling of three-dimensional and shell parts in the same structure, the piezoelectric driving force, the fluid–structure interaction as well as the radiation damping. This paper describes the analysis of a test axisymmetric transducer with the ATILA code [J. N. Decarpigny *et al.*, J. Acoust. Soc. Am. **78**, 1499 (1985)] using dipolar damping elements and a new extrapolation method to obtain the transducer farfield characteristics [R. Bossut *et al.*, J. Acoust. Soc. Am. Suppl. 1 **79**, S51 (1986)]. In-air resonance modes, transmitting voltage response and directivity patterns are computed and compared to measurements, displaying a satisfactory agreement. Finally, the modeling of a transducer that is built with a glass-reinforced plastic shell is described, and the corresponding problems and results are discussed. [a] Currently at Thomson-Sintra, Chemin des travails, 06802 Cagnes sur mer, France.

2:30

L5. Array shape estimation via piecewise subarray beamforming. Deanna M. Caveny, Donald R. Del Balzo, Jeffrey L. Becklehimer (NORDA, Code 244, NSTL, MS 39529), and George E. Ioup (Department of Physics and Geophysical Research Laboratory, University of New Orleans, New Orleans, LA 70148)

Deformation produced when towing a linear array can remove L–R ambiguities in conventional beam patterns, provided the beamformer knows the deformed array shape. In practice, however, this shape is usually unknown. Thus it is desirable to have a processor that determines the shape and uses this information to form the beams. A method which accomplishes this goal forms beams for subapertures separately assuming that they are approximately linear. A maximum likelihood estimation method [M. J. Hinich and W. Rule, J. Acoust. Soc. Am. **58**, 1023–1029 (1975)] has previously been used to associate an angle with each of the subapertures. Given the subarray bearing angle they calculated a mean and a median to estimate the full array bearing angle. In this study, after the main angle was determined, using conventional methods, a piecewise linear array was constructed by placing the subarrays together with estimated correction angles. To test the performance of the approximated array shapes, beamforming was carried out using the new piecewise linear array and the modeled data for the original deformed array. Systematic errors were investigated as well as various methods for smoothing the piecewise linear array.

2:45

L6. Performance of sinusoidally deformed line arrays. Deanna M. Caveny, Donald R. Del Balzo, Jeffrey L. Becklehimer (NORDA, Code 244, NSTL, MS 39529), and George E. Ioup (Department of Physics and Geophysical Research Laboratory, University of New Orleans, New Orleans, LA 70148)

Previously it has been shown [M. J. Hinich and W. Rule, J. Acoust. Soc. Am. **58**, 1023–1029 (1975); W. S. Hodgkiss, IEEE J. Ocean. Eng. **OE-8**, 120–130 (1983)] that deformations of towed arrays from a straight line shape can produce significant distortions in beam patterns and errors in bearing estimation if the beamforming assumes linearity. It has also been shown that a deformed array helps to remove left/right ambiguities in the beam patterns, provided the beamforming is done with the correct array configuration. In this work these two effects are studied for undamped and damped sinusoidally deformed arrays (as observed in practice) of one, two, and three half-cycles with relatively small array amplitudes. By use of fixed arc length separations along the array, the phone (x, y) coordinates are determined numerically for each sinusoidal shape. The complex pressure fields are modeled for sources at various locations. Then beamforming is carried out (1) with the known array configuration, and (2) assuming that the array is linear. Degradations resulting from assuming linearity and the ability to remove left/right ambiguities are discussed in terms of reduced gain, angular resolution, and bearing errors.

3:00

L7. On the design technology of the uncompensated class IV flextensional transducer. G. A. Brigham (Aquasonics, Inc., Anaheim, CA 92806)

The air-backed flextensional transducer is stress constrained both in the shell and the electromechanical bar driver. When Navy type III hard lead zirconate–titanate ceramic is used, the total amount of bar preload for depth and power cannot exceed 10 kpsi with allocation an interative process in the design. The math model consists of three radiating shell modes together with the bar end velocity at the shell–bar interface. The design starts with a stress analysis of shell and bar which yields the relative dimensions of both. These are input to (a) the shell–bar equation for mechanical resonance in water to get the shell major axis width, and (b) the resonant half-power bandwidth equation to get the shell length. All remaining parameters follow from the sizing, e.g., effective coupling, weight, and peak acoustic power. This paper shows the design sequence, the results for a uniform elliptic ring used as a baseline, and application to designing various flextensionals for several new major Navy sonar transmit array programs.

3:15

L8. Fringe counting demodulator for fiber optic interferometric sensors. C. M. Crooker and S. L. Garrett (Physics Department, Code 61 Gx, Naval Postgraduate School, Monterey, CA 93943)

A demodulation scheme for high sensitivity (1–10 krad regime) fiber optic interferometric sensors which is based on fringe rate has been developed. The technique is similar to that utilized in optical shaft encoders.

The output of a shaft encoder driven by a pendulum resembles the fringe pattern produced by a fiber optic interferometric sensor in response to a sinusoidal perturbation. This similarity was exploited to design and calibrate the demodulation circuit. An F-to-V chip first converts the shaft rotation rate to a proportional voltage level. Additional circuitry determines whether the voltage level should remain positive or be inverted through a comparison of the in-phase and quadrature waveforms output by the shaft encoder. A 3×3 coupler provides these waveforms optically [S. Sheem, J. Appl. Phys. **52**, 3865 (1981)]. [Work supported by the Naval Research Laboratory and the Office of Naval Research.]

3:30

L9. Effect of a voltage shading of the ceramic stack on the bandwidth of a radiating Tonpilz transducer. D. Boucher, B. Tocquet[a] (Groupe d'Etude et de Recherche de Détection Sous-Marine, Le Brusc, 83140 Six Fours les Plages, France), J. N. Decarpigny, J. C. Debus, and P. Tierce (Institut Supérieur d'Electronique du Nord, 41 boulevard Vauban, 59046 Lille Cédex, France)

In addition to the main resonance, the analysis of classical Tonpilz transducers points out the existence of several maxima of the transmitting voltage response which are associated to various structural modes (dilational modes of the stack, flexural modes of the headmass). However, though the electromechanical coupling of the first mode is large, the other modes are generally less effective. In this paper, the effectiveness of a particular mode is shown to depend upon a simple relation, for each ceramic ring of the stack, between the phase of its electrical excitation and the sign of its longitudinal strain. This fact is clearly demonstrated with the help of a finite element modeling, using the ATILA code [J. N. Decarpigny *et al.*, J. Acoust. Soc. Am. **78**, 1499 (1985)]. These results suggest various voltage shadings of the ceramic stack which reinforce particular modes. Thus various frequency ranges can be effectively obtained with the same transducer, as confirmed by a detailed study that is based on numerical simulations as well as in-water measurements. [a] Currently at Thomson-Sintra, Chemin des travails, 06802 Cagnes sur mer, France.

3:45

L10. Automated in-service test and evaluation of sonar hydrophones. Richard E. Self, J. D. Johnson, O. B. Wilson, and S. L. Garrett (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

Recent advances in microprocessor-based instrumentation permit rapid, precise, and detailed measurements on sonar transducers in the field. Reliable data combined with equivalent circuit models of the transducer provide information on acoustical sensor performance that can help to establish more realistic performance and maintenance criteria. This can substantially upgrade the quality of maintenance decisions made at the technician level in the field in light of the effects on well-understood parameters that relate directly to operational performance, such as sensitivity, array beam width, and bearing error, due to transducer degradation. Transducer complex admittance measurements made in-the-field using a commercially available computer controlled impedance meter will be described. Interpretation of the data and their relationship to the equivalent circuit model will be discussed and typical results will be presented. [Work supported by the Naval Sea Systems Command.]

4:00

L11. Automated *in situ* measurement of hydrophone sensitivity for hull-mounted submarine sonar arrays using a reciprocity method. Gibson B. Kerr, Patricia M. O'Neill, O. B. Wilson, and S. R. Baker (Department of Physics, Code 61WL, Naval Postgraduate School, Monterey, CA 93943)

The United States Submarine Force currently conducts lengthy pre- and post-repair evaluation of submarine hull-mounted arrays using extensive, highly specialized external equipment and facilities. A methodology using transmission of acoustic energy between array elements employing portable test equipment controlled by a desktop computer has been developed that allows very rapid measurement of voltage and current parameters required for calculating sensitivities of acoustic transducers *in situ*. The method is an adaptation of one described by Van Buren [NRL Memorandum Report 5642, July 1985]. The method has been successfully applied to the determination of relative sensitivities for staves of DT-276 hydrophones in submarine hull-mounted arrays. Also, the method has been successfully adapted to detect the improper wiring of hydrophones in a stave of the array. The method is valuable since all measurements are performed from onboard the submarine. An outline of the procedures and examples of results will be presented. [Work supported by Naval Sea Systems Command.]

4:15

L12. An evanescent wave array for wavenumber calibration. H. C. Schau, L. Dwight Luker, S. Petrie, and A. L. Van Buren (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337)

A transmitting array that is designed to generate specified evanescent pressure fields near the array is described. The pressure fields are convective in nature, moving parallel to the plane of the array with a phase velocity that is slow, i.e., less than the speed of sound in water. A slow phase velocity corresponds to a spatial wavenumber k which is greater than the corresponding acoustic wavenumber in water. Such an array can be used to obtain the response of extended acoustic sensors to high-wavenumber pressure components such as exist in turbulent flow fields. Two small prototype planar arrays have been designed and constructed. The first one utilizes the piezoelectric polymer (PVDF) as its active material while the other one utilizes a Neoprene–lead titanate composite material. The positive electrodes on both arrays are segmented into independently driven areas by use of equally spaced stripes. Experimental results obtained for the prototype arrays are compared with computer model predictions. [Work partially supported by ONT.]

4:30

L13. Planar turbulent boundary layer (TBL) pressure field emulation with a reduced degree of freedom array. H. C. Schau (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856-8337)

Planar arrays being developed for producing one-dimensional convective pressure fields with prescribed spatial wavenumber content, slow phase velocity, and temporal frequency are described in the preceding paper. Such devices are capable of "frozen" turbulence emulation but allow no specified correlation properties such as those assumed to exist beneath Turbulent Boundary Layers (TBL). Extending the one-dimensional technique (involving N independently driven stripes) to two dimensions (with N^2 independently driven rectangular areas) becomes unfeasible as the array size becomes large due to the extremely high bandwidth required. In this paper, an alternate approach is proposed and the trade-off between performance and implementation is discussed. The conclusions reached in this study include the types of TBL second order statistics which the device can produce within implementation constraints and methodologies for calibration of extended area sensors to TBL noise in a repeatable fashion.

4:45

L14. The effect of electrode stiffness on the piezoelectric and elastic constants of a piezoelectric bar. Mark B. Moffett (Naval Underwater Systems Center, New London, CT 06320) and Donald Ricketts (Raytheon Company, Portsmouth, RI 02871)

Electrodes that are stiff and thick compared with the underlying piezoelectric substrate material can substantially change the effective piezoelectric and elastic constants from those values which would be obtained in the absence of electrodes. A simple analytical model for the stress distribution inside a composite piezoelectric plate, consisting of two outer electrode layers and one inner piezoelectric layer, is used to calcu-

TUESDAY PM

late the effective g_{31}, g_{32}, and g_{33} piezoelectric constants, as well as the effective s_{ij} $(i,j = 1,2,3)$ elastic compliance coefficients. Numerical results for copper-clad poly(vinylidene) fluoride (PVDF) plates are presented. [Work supported by the Office of Naval Technology and by Raytheon Submarine Signal Division, Acoustic Transduction Research and Development Project (DR).]

TUESDAY AFTERNOON, 9 DECEMBER 1986 — SALONS J AND K, 1:30 TO 4:20 P.M.

Session M. Noise III: Noise from Wind-Energy Farms

T. James DuBois, Chairman
Southern California Edison, P. O. Box 800, Rosemead, California 91770

Harvey H. Hubbard, Co-Chairman
NASA Langley Research Center, MS 463, Hampton, Virginia 23665-5225

Chairman's Introduction—1:30

Invited Papers

1:35

M1. Wind turbine technology—not as simple as it looks. Michael C. Wehrey (Southern California Edison Company, Rosemead, CA 91770)

Wind is a clean and inexhaustible energy resource serving mankind for many centuries by driving windmills to grind grain and pump water. This presentation gives a brief historical review from the earliest drag-type windmills existing as early as 2000 BC through the early electricity-generating units in the early 1900s to the present-day wind turbine parks. The 1973 oil embargo and 1979-80 price increases brought new awareness of conservation and promoted new interest in renewable energy resources and wind turbine technology. Many lessons are being learned in the design of modern wind turbines. The quest for low installation and maintenance costs, energy conversion efficiency, and high reliability continues. Unforeseen environmental issues such as visual pollution, noise impacts, and TV reception interference are to be addressed. The technical features and operating characteristics of various designs are presented including problems encountered and their solutions.

2:05

M2. Acoustic signatures of five wind turbine generators from 80–500 kW. T. James DuBois (Southern California Edison Company, Rosemead, CA 91770)

As part of its research and development program for development of renewable energy sources, Southern California Edison has monitored the acoustic emission of several different horizontal and vertical axis wind turbine generators with power generation capacity ranging from 80–500 kW. Noise measurements included: frequency spectrum (infrasonic through audible range), directivity, nearfield and farfield, and short and long duration. This presentation will summarize the results of the tests and measurements of both vertical and horizontal axis wind turbines.

2:35

M3. Results of simultaneous acoustic measurements around a large horizontal axis wind turbine. Harvey H. Hubbard and Kevin P. Shepherd (The Bionetics Corporation, Hampton, VA 23666) and William L. Willshire (NASA Langley Research Center, Hampton, VA 23665)

A unique set of experiments is described in which simultaneous acoustic measurements were made at 18 locations around a large horizontal axis wind turbine in a circular array having a radius of 200 m. Results are presented for frequencies from 3–3000 Hz, for power output values from 0.1–2.1 MW and for several rotational speeds and skew angles. Additional data for simultaneous measurements using a linear array will be presented to illustrate sound propagation in both the upwind and downwind directions to distances of several kilometers. [Work supported by DOE.]

3:05

M4. Signal processing for windmill noise measurement. Bruce Walker (Consulting & Research in Acoustics, 2659 Townsgate Road, Suite 101, Westlake Village, CA 91361)

Measurement of noise from wind-powered generators (windmills) often must be done in acoustically hostile environments, with ambient winds of up to 15m/s and operation of large numbers of noise sources (other windmills, highways, etc.) in the same vicinity. In addition, the frequency spectra produced by some commonly encountered windmills extend well into the infrasonic range, where microphone/wind-interaction noise is most troublesome. By using autocorrelation and cross-correlation processing and homodyne filtering, it has been possible to resolve windmill noise data from signal-to-noise environments as poor as 15 dB with good repeatability. The paper will discuss how these processing techniques have been implemented on a desk top computer with a minimum of expense. The paper will also include an audio demonstration of the noise characteristics of a variety of windmills.

Contributed Papers

3:35

M5. Wind-energy farm noise impacts on nearby residences—the citizen's view. John R. Warner (P. O. Box 387, North Palm Springs, CA 92258)

Wind-energy farms, with hundreds of wind turbines, have proliferated in close proximity to residential properties, without any apparent control on the noise impact consequences. Annoying and intrusive wind turbine noise occurred as soon as the first turbines were installed. The adverse noise impact has continued and intensified as additional WECS (wind energy conversion system) sites were approved and developed with the installation of hundreds of additional turbines. The adverse noise effects consist of low-frequency noise that is not fully addressed or controlled by only A-weighted sound level standards, pure tones, and higher noise levels during higher wind-speed conditions. More stringent noise controls should have been implemented in the beginning. Riverside County has recently adopted a more stringent noise standard that also includes a pure tone restriction that will apply to future sites. However, mitigation of the mistakes made at the earlier sites remains to be resolved.

3:50

M6. Evaluation of wind turbine noise levels and impact studies. Samuel R. Lane (2044 Swan Drive, Costa Mesa, CA 92626)

Measured A-weighted sound levels at 125-ft distance for individual wind turbines with 20- to 120-kW power ratings are typically in the range 65–75 dB at moderate to high power output conditions (20–30-mph wind speeds). Tonelike sounds in the 300- to 1000-Hz frequency range often are clearly audible. Cyclical fluctuations of 10 dB in low-frequency noise levels are propagated by some downwind-type turbines. The random aerodynamic rotor noise sounds like a roar, the gear box noise sounds like a whine, and the low-frequency noise fluctuations sound like "thump–thump" or "whoosh–whoosh." All of these wind turbine noises are propagated from existing wind farms to residential areas and are judged intrusive and annoying. Measurements and predictions of wind turbine noise submitted with applications for wind farm development have often contained errors which understated the noise levels by 3–10 dB. These errors were due to noise measurements at minimal wind speeds and turbine power, and faulty modeling procedures. Simple analytical expressions have been developed which quickly and accurately predict the noise levels for large turbine arrays.

4:05

M7. Criteria for low frequency and infrasound from wind-energy farms. Jim Buntin and Robert E. Brown (Brown–Buntin Associates, 5908 Fair Oaks Boulevard, Carmichael, CA 95608)

Criteria for low frequency and infrasound were developed under contract to Kern County, California, to allow assessment of potential public reaction to wind-energy farm developments. Recent literature was reviewed regarding possible annoyance, physical discomfort, and perception of sound in the frequency range of 2–125 Hz. Criteria suggested by a number of authors were compared to determine commonality of recommended sound pressure levels versus frequency in one-third octave bands. The suggested criteria were than evaluated as to whether they addressed the issue of annoyance, rather than perception. A set of criteria was then proposed which incorporated empirically derived criteria intended to minimize community reaction, incorporating a 5-dB penalty for the impulsive components which are sometimes present in low frequency and infrasound produced by wind-energy farm installations. The resulting criteria represent a balance between sound pressure levels which are perceptible and those which are expected to result in community annoyance.

TUESDAY AFTERNOON, 9 DECEMBER 1986 — SALON F, 1:30 TO 4:00 P.M.

Session N. Psychological and Physiological Acoustics II: Cochlear Implants, Hearing Aids, and Product Development

William McFarland, Chairman
Otologic Medical Group, 2122 West 3rd, Los Angeles, California 90057

Contributed Papers

1:30

N1. Forward masking studies on a multichannel cochlear implant patient. H. H. Lim, Y. C. Tong, G. M. Clark, and P. A. Busby (Department of Otolaryngology, Melbourne University, Parkville, 3052, Australia)

The profile of neural excitation pattern along the cochlea produced by a longitudinally oriented bipolar electrode pair was estimated by forward masking. Two types of masking patterns were observed. The first was produced by electrode pairs with low (behavorial) threshold currents, and displayed two peaks approximately located above the two electrodes of the bipolar pair. The second was produced by pairs with higher threshold, and displayed a single peak approximately midway between the two electrodes. These masking patterns are similar to the two types of current distribution patterns observed in a tank model consisting of a 22-banded electrode array fixed into a normal saline filled tube of 5 mm diameter. It is possible that the bipolar pairs with lower threshold and double-peaked masking patterns are situated closer to the residual auditory fibers and neurons. In another study, the masking patterns for two bipolar electrode pairs interleavely stimulated were found to follow approximately the maximum of the two masking patterns for the two electrode pairs activated in isolation.

1:45

N2. Modeling studies on current distributions produced by an intracochlear electrode array. P. M. Lukies, Y. C. Tong, G. M. Clark, and P. A. Busby (Department of Otolaryngology, University of Melbourne, Parkville, 3052, Australia)

Modeling studies on current distributions along the cochlea produced by an intracochlear electrode array were conducted in a tank model. The tank consisted of a 22-banded electrode array fixed into a normal saline-filled tube of 5 mm diameter. Bipolar electrical stimulation was used, and the potentials within the saline were measured to enable the calculation of the current distribution. Current distribution depended on the ratio of the radial distance from the electrode array at which the distribution was measured to the distance between the two electrode bands used for bipolar stimulation. For small ratios, the distribution displayed two peaks directly above the two bands; while for large ratios, the distribution displayed only one peak midway between the two bands. Also investigated was the interaction that occurs when more than one bipolar pair were stimulated simultaneously. The results showed that the resulting distribution also depended on the above defined ratio. These results obtained from the tank model were verified by computer simulation.

2:00

N3. Reduction of interaction in multichannel cochlear implants. Brent Townshend (Stanford Electronics Laboratory, AEL 136, Stanford University, Stanford, CA 94305)

The spread of current in multichannel implants results in excitation of largely identical neural populations regardless of which electrode is stimulated. This severely reduces the number of independent channels of information conveyed with monopolar or even bipolar stimulation. A method based on the thresholds of multiple electrode stimuli was developed to calculate accurately the mapping from electrode current levels to normalized current density at each neural site. For the implant patients tested, this technique accurately predicted thresholds for arbitrary patterns of stimulation. Inverting this mapping matrix produced patterns of stimulation that selectively stimulated only one neural site with a minimum of interaction, while satisfying current limit constraints. These *sharpened* stimuli were pitch ranked with greater consistency than either monopolar or bipolar stimuli. [Work supported by NIH.]

2:15

N4. Psychophysical studies on prelingual patients using a multiple electrode cochlear implant. Y. C. Tong, P. A. Busby, and G. M. Clark (Department of Otolaryngology, Melbourne University, Parkville, 3025, Australia)

Psychophysical studies were conducted on three prelingual patients during the first year of usage of a multiple electrode cochlear implant. The results were compared with those obtained from postlingual patients. For the identification of two electrode positions (6 mm apart), the d' sensitivity index for prelinguals was <0.75 and for postlinguals >3.0. For the identification of two electric repetition rates (an octave apart), d' for prelinguals was <0.6 and >3.0 for postlinguals. Temporal resolution discrimination was also poorer for prelinguals, gap detection >64 ms and duration difference limens >38 ms from a fixed standard duration of 30 ms, while for postlinguals, <3 ms and <38 ms, respectively. The ability to count a variable number of short electrical stimuli deteriorated at high rates (>5/s) of presentation for prelinguals but not for postlinguals, while for visual and tactile stimuli performance was similar across all patients. However, improvements in one prelingual patient after one year of usage were recorded for electrode place and rate identification ($d'>1.5$).

2:30

N5. Speech perception studies in the first year of usage of a multiple electrode cochlear implant by prelingual patients. P. A. Busby, Y. C. Tong, and G. M. Clark (Department of Otolaryngology, Melbourne University, Parkville, 3052, Australia)

Vowel and consonant confusion studies, in the electrical alone (E), visual alone (V), and electrical plus visual (EV) conditions, and three Minimal Auditory Capabilities (MAC) subtests, in the E condition, were conducted on three prelingual patients using a multiple electrode cochlear implant. For vowels, the one-dimensional solution from multidimensional scaling analysis for the E condition was interpreted as vowel length, and the one-dimensional and two-dimensional solutions for the V and EV conditions as visual parameters. The constant results from information transmission analysis for the E condition indicated poor transmission of all articulatory features, and minimal differences between transmission scores for the V and EV conditions were recorded. For the MAC subtests, two patients scored significantly above chance for male–female speaker identification, one patient for one syllable–two syllable identification, but no patients for spondee same–different discrimination. The results suggested that these prelingual patients did not fully utilize the information provided by a multiple electrode implant during the first year of use.

2:45

N6. Cochlear neuronal degeneration precipitated by experimental blockage of the spiral lamina perforated zone. Hugh S. Lusted (Division of Otolaryngology, R-123, Stanford University School of Medicine, Stanford, CA 94305)

Observation of patterns of neuronal degeneration in cats with cochlear implants has led to the hypothesis that blockage of the spiral lamina perforations (*canaliculae perforantes* of Schuknecht) by the implant produces localized dendritic degeneration and ganglion cell loss. To test this hypothesis four cats were implanted with 1×1 mm squares of polyethylene (PE) film placed directly on the perforated zone (four ears). Four control ears received similar PE squares placed on the scala tympani wall opposite the basilar membrane. Histological examination of the cochleae after 3 months survival time revealed almost total dendritic degeneration and 50%–75% ganglion cell loss in localized regions of the cochlea where the implant fibrous encapsulation could be seen in direct contact with the perforated zone. No neuronal degeneration was observed in the cochleas which showed no fibrous tissue in contact with the perforated zone. These results indicate that the perforated zone is a contact critical area and needs to be avoided by implanted electrodes intended for functional stimulation of the cochlea. [Work supported by NIH.]

3:00

N7. Processor controlled hearing aid. Samuel Gilman (Sam Gilman Assoc., P.O. Box 25176, West Los Angeles, CA 90025)

The amplified signal input from the hearing-aid microphone is split into one-third octave bands, each having its own processor-controlled band amplilfier. Control signals are obtained from a feedback probe microphone at the medial end of the hearing aid earmold. This signal is analyzed into one-third octaves, digitized, and compared to stored band levels in the processor (as obtained from the audiologically specified eardrum SPL and spectrum). The gain of each band amplifier is controlled by the processor so that the spectrum and SPL at the eardrum is the same as the stored spectrum and SPL regardless of variations in input signals, hearing aid characteristics, and external ear acoustics. Processor control of averaging, attack, and release times have been provided. Preliminary

ANSI S3.25-1979 simulator graphic and listening tests show that band levels and averages at the eardrum are maintained within a few dB for variations of over 30 dB in the input spectrum and level.

3:15

N8. Using multiple microphones and LMS adaptive beamforming to reduce interference in hearing aids from competing talkers in reverberant rooms. Patrick M. Peterson (MIT Research Laboratory of Electronics, Room 36-758, Cambridge, MA 02139)

To reduce interference in monoaural hearing aids from spatially separated sound sources, the information from multiple microphones might be useful in enhancing the signal from a desired source for monaural presentation. The technique of Griffiths and Jim [IEEE Trans. Antennas Propagat. **AP-30**, 27–34 (1982)] was used to filter and combine microphone signals in a way that preserves signals arriving from straight ahead of the microphone array while minimizing output power from off-axis signals. The filters adapt to the interference environment using a variation of the LMS (Widrow) algorithm. Although it adapts slowly (1 s), LMS approaches optimum performance in stationary, anechoic environments, providing a bound on performance in nonstationary, reverberant environments. In three simulated rooms, anechoic, living room, and conference room, with interferring babble 45° off-axis and on-axis target at −12 dB/S/B, a two-microphone system with 100-point filters reduced the interference by 20, 12, and 4 dB, respectively. Informal listening indicated minimal degradation of the desired signal and substantial intelligibility improvement in anechoic and living room environments. [Work supported by NIH.]

3:30

N9. Robust transformation of middle ear impedance parameters from an intra-aural measuring plane to the eardrum. J. A. del Rio, H. Kunov, R. K. Hong, and P. Malone (Institute of Biomedical Engineering, University of Toronto, Toronto, Ontario M5S 1A4, Canada)

In order to eliminate the risk inherent in the insertion of a measuring probe into the ear canal of a subject, a middle ear impedance transducer should be placed a safe distance away from the eardrum. A technique presented in this paper uses a function which represents the area function of the ear canal and transforms impedance parameters from the measurement plane to a plane at the eardrum. The effects of the geometry of the canal on impedance measurements, and the validity of the approximations over a wide frequency range are studied by means of a computer model. The area functions used to transform the impedance parameters are parabolic horns which are found to provide better results than the commonly used cylindrical approximation. The sensitivity of these approximations to errors in the estimation of the parameters defining the horn is also studied. The parabolic horns give a simple and very reliable correction to the interpretation of middle ear impedance data. [Work supported by the Natural Sciences and Engineering Research Council of Canada, CRD-8422.]

3:45

N10. ISPUD, an interactive environment for signal processing on a VAX computer. Patrick M. Peterson and Joseph A. Frisbie (MIT Research Laboratory of Electronics, 50 Vassar Street, Room 36-758, Cambridge, MA 02139)

To support our group's research in psychoacoustics and hearing aids, a useful software package, ISPUD, for computer-based signal processing has been developed. Following the work of Kopec [IEEE Trans. Acoust. Speech, Signal Process. **ASSP-32**, 842–851 (1984)], ISPUD uses abstract data objects to represent signals in a way that leads to a simple, interactive, and extensible environment for signal processing. In contrast to Kopec's package, which requires a LISP machine and supports one user, ISPUD runs on a VAX with the multi-user VMS operating system, a C compiler, and Tektronix-4010 compatible terminals. The user interface is a simple, LISP-like interpreter for calling defined functions or combining functions into new definitions. The package includes efficient built-in functions, written in C, for signal generation, arithmetic, composition, windowing, filtering, transforms, statistics, graphical display, and file I/O. In addition, user-written C functions can be incorporated. Examples of ISPUD usage, including strengths and weaknesses, will be presented. [Work partially supported by NIH.]

TUESDAY AFTERNOON, 9 DECEMBER 1986 — SALONS A AND B, 1:30 TO 2:30 P.M.

Session O. Shock and Vibration II: Distinguished Lecture on Active Control of Structural Vibration

Sabih I. Hayek, Chairman
Department of Engineering Science and Mechanics, Pennsylvania State University, University Park, Pennsylvania 16802

Invited Paper

1:30

O1. Active control of structural vibration. Leonard Meirovitch (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

In general, structures represent distributed-parameter systems described by partial differential equations. Strictly speaking, vibration control in distributed structures requires distributed sensors and actuators. Practical considerations demand that control be carried out by means of discrete components. Quite often they also demand that the structure be discretized in space. A discretization procedure having particular appeal is the finite element method, which has the advantage that the nodal (in the finite element sense) coordinates represent actual displacements of the structure. One of the most significant problems in feedback control is the generation of control gains. Techniques for computing gains in common use are the pole allocation method and optimal control. These methods work best for low-order systems. This tends to create a problem in structures, which generally demand high-order mathematical models. Feedback control requires determination of the system state, which can be done through measurement of the nodal displacements and velocities. If it is not feasible to measure all the components of the state, then it is possible to construct a dynamical system, related to the actual system and known as a Luenberger observer, to estimate the full state. One method of controlling the vibration of a structure is modal control, which implies controlling the modes of vibration of the structure. In the independent modal-space control method, design of the control can be carried out for each mode independently. This requires estimation of the modal states. A method not requiring estimation of the modal states is direct feedback control, in which the sensor signals are amplified directly to generate the feedback forces. Under given circumstances, direct feedback can be rendered nearly optimal.

TUESDAY AFTERNOON, 9 DECEMBER 1986 — SALONS A AND B, 2:30 TO 5:05 P.M.

Session P. Shock and Vibration III: Radiation Loading on Elastic Structures

Miguel C. Junger, Chairman
Cambridge Acoustical Associates, Inc., 54 Cambridge Park Drive, Cambridge, Massachusetts 02140

Invited Papers

2:30

P1. Asymptotic solution to low-frequency fluid-loaded structure problems. D. G. Crighton (D.A.M.T.P., Cambridge University, Silver Street, Cambridge CB3 9EW, England)

This paper will review the structural and acoustical features of asymptotic solutions recently obtained for fluid-loaded structure problems. In the cases treated here, the structure is plane (a thin plate or membrane) and driven by a source of concentrated mechanical excitation. The asymptotic results, based on the smallness of the fluid-loading-at-coincidence parameter, hold generally at frequencies below coincidence, but assume particularly simple forms at much lower frequencies. Problems discussed will include semi-infinite and finite unribbed structures (with details of resonance, mode shapes, and forced response), and infinite and semi-infinite ribbed structures. An aspect subject to particular study will be the nonlocal (i.e., non-added-mass) effects of fluid loading; and generally, the aim is to expose fundamental mechanisms for structural vibration with heavy fluid loading, and to provide benchmarks for computational schemes. [Work supported by ONR.]

2:55

P2. Fluid-loading effects upon scatter of elastic waves at discontinuities. P. W. Smith, Jr. (BBN Laboratories, Inc., 50 Moulton Street, Cambridge, MA 02138)

Vibratory waves in structures constructed of many plate or shell elements joined together are scattered by the discontinuity of elastic properties at the junctions. Descriptions of that scatter (reflection, transmission, and radiation into adjacent fluid) are important in analyzing the behavior of realistically complex structures. Results of analyses that have included significant fluid loading are reviewed and their implications for structural response discussed.

3:20

P3. Finite element treatment of exterior fluids in fluid–structure interaction problems. Gordon C. Everstine [Numerical Mechanics Division (184), David Taylor Naval Ship Research and Development Center, Bethesda, MD 20084]

Several finite element solution approaches are described for fluid–structure interaction problems involving an exterior fluid. The specific problems of interest are those of acoustic radiation and scattering from submerged elastic structures subjected, respectively, to internal time-harmonic mechanical loads or an incident time-harmonic wavetrain. These problems can be solved by combining a finite element model of the structure with a fluid loading computed using finite element, boundary element, infinite element, or decoupling techniques. Formulations are presented and compared for fluid domains modeled using an added mass matrix, the doubly asymptotic approximation (a decoupling technique), explicit fluid finite elements, and the Helmholtz exterior integral equation. With the exception of the integral equation treatment, all these fluid-loading approaches can be implemented using the standard finite element modeling capabilities available in structural analysis computer codes such as NASTRAN. The most accurate fluid treatment is obtained with the Helmholtz integral equation although this is at somewhat greater computational cost than that required using the less precise approaches.

3:45

P4. Doubly asymptotic approximations in acoustic, elastodynamic, and electromagnetic scattering. Thomas L. Geers (Department of Mechanical Engineering, University of Colorado, Boulder, CO 80309)

Doubly asymptotic approximations (DDAs) are differential equations that describe the dynamic interaction between a scatterer and a surrounding wave-propagating medium. They find application in problems of acoustic, elastodynamic, and electromagnetic scattering by bodies of complex geometry and structure through the utilization of boundary element and finite element techniques. DAAs have been successfully developed and applied for acoustic scattering, have been developed for elastodynamic scattering, and are under development for electromagnetic scattering. In this paper, systematic formulations of DAAs for the various field equations are presented, and their behavior in canonical problems is examined. The utilization of increasingly higher-order DAAs is explored and the point of diminishing returns is sought.

4:10–4:35

Round Table Discussion: Unsolved Problems

M. C. Junger, Chairman
Cambridge Acoustical Associates, Inc., 54 Cambridge Park Drive, Cambridge, Massachusetts 02140

The discussion is intended to identify important unsolved problems, to define priorities, and to compare alternative approaches.

Contributed Papers

4:35

P5. Noise transmissibility of mechanical systems in a submerged elastic shell. H. Huang (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20903-5000) and Y. F. Wang (David W. Taylor Naval Ship Research and Development Center, Annapolis Laboratory, Annapolis, MD 21402-1198)

The effects of fluid–structure interaction on the force transmissibility of mechanical systems elastically mounted in a submerged elastic shell are investigated. An analytical solution for the case of a harmonic-force-excited two-degrees-of-freedom undamped system attached inside a spherical elastic shell is obtained by the separation of variable technique. The transmissibility of the internal system is computed for a broad range of excitation frequencies and compared with that of the same system mounted on a fixed base. Due to the loss of energy through the radiation of the elastic shell, the transmissibility resonance peaks of the undamped system elastically mounted in the submerged shell are reduced to finite values and, for other nonresonance frequencies, the magnitude is in general also reduced. Moreover, the fluid–structure interaction effect shifts the peaks of the transmissibility towards lower frequencies. This information could be useful for the sound isolation of systems inside a submerged vessel.

4:50

P6. Experimental investigation of fluid-coupled mechanical waves propagating on cylindrical shells at low *ka*. Bernard Garnier and Bernard Vernozy (Métravib R.D.S., B.P. 182, 69132 Ecully Cédex, France)

A set of acoustical arrays has been constructed, the shapes of which are configurated in accordance with the cylindrical geometry of the near-field of the shells (12–20 in.-diameter). Some are double layer arrays, and directly give the acoustic intensity of radiated noise through the array. Other give the dynamic pressure in the immediate vicinity of the shell, and reflect both propagating and nonpropagating fields, like those associated with subsonic flexural waves. The studied shells reflect a wide assortment of usual shell technologies in torpedoes or ship building: steel, aluminum alloys, glass-, or carbon-fiber reinforced resins; ribbed shells, more or less disturbed by internal fittings, etc. They are mechanically excited by various arrangements of piezoelectric shakers driven by recurrent pulses. When the measurements are plotted as time/space recordings, they give a descriptive image of wave propagating in the shells: phase and group velocities, damping, and scattering on all discontinuities of the shell, which is generally the main phenomenon to explain the effective propagation of vibrations along realistic shiplike structures.

TUESDAY AFTERNOON, 9 DECEMBER 1986 — SALON E, 1:30 TO 5:00 P.M.

Session Q. Speech Communication II: Vector Quantization and its Applications to Speech Research

Biing-Hwang Juang, Chairman
AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, New Jersey 07974

Invited Papers

1:30

Q1. Vector quantization for speech coding and recognition. Robert M. Gray (Electrical Engineering Department, Stanford University, Stanford, CA 94305)

Vector quantization (VQ) is the coding or mapping of continuous or high-resolution vectors into a finite rate digital representation. The general goal is either analog-to-digital conversion or data compression with a minimum loss of fidelity. In this paper, the basic vector quantizer structures and code design techniques for fixed rate vector quantizers are surveyed and applications to speech coding and recognition are described.

2:00

Q2. Relative entropy and vector quantization. John E. Shore (Entropic Processing, Inc., Washington Research Laboratory, 600 Pennsylvania Avenue S.E., Suite 202, Washington, DC 20003)

Consider the problem of quantizing a vector F of measurements F_r, $r=0,\ldots,M$. In one general form of vector quantization F is quantized as the result of classification by a nearest-neighbor rule,

$$D(\mathbf{F},\widehat{\mathbf{F}}^{(t)}) = \min_{s\in\Lambda} D(\mathbf{F},\widehat{\mathbf{F}}^{(s)}),$$

where D is some distortion measure and where $\{\mathbf{F}^{(s)}:s\in\Lambda\}$ is a set of predefined vectors, often called codewords. Often, the measurements F_r can be expressed as a set of expected values, $\int q^\dagger(\mathbf{x}) f_r(\mathbf{x})d\mathbf{x} = F_r$, and each codeword $\mathbf{F}^{(s)}$ can be expressed similarly. In such cases, the *relative-entropy*, $H(q,p) = \int q(\mathbf{x})\log[q(\mathbf{x})/p(\mathbf{x})]d\mathbf{x}$, can be used to define a distortion measure in (1) so that the classification is optimal in a well-defined information-theoretic sense and also computationally attractive. Furthermore, the distortion measure results in a simple method of computing codewords from training data. The method of speech coding by vector quantization is a special case of this relative-entropy method.

2:30

Q3. Matrix quantization for very low data rate voice coding. David Y. Wong (Digital Sound Corporation, Santa Barbara, CA 93103), B. H. Juang (AT&T Bell Laboratories, Murray Hill, NJ 07974), and D. Y. Chen (Digital Sound Corporation, Santa Barbara, CA 93103)

Matrix quantization extends the theory of vector quantization for voice coding to take advantage of the phonological and phonotactic properties of speech. The speech signal is first modeled by LPC spectra at 10-ms time intervals. By adopting an acoustically defined (and demisyllable like) automatic segmentation procedure, consecutive LPC spectra are grouped into LPC coefficient matrices. A distortion measure that consists of nonlinear time warping and a standard LPC distance metric is also developed for comparing the "likeliness" of two matrices. The matrix codebook is generated from a speech database using a simple mini-max procedure and the matrix distortion measure. A nearest-neighbor full search procedure is used during speech encoding. Data rates under 100 bit/s can be achieved for the LPC spectra. Results for a single speaker experiment based on the matrix quantization method and subsequent research on very similar techniques will be discussed.

3:00

Q4. A talker recognition based on vector quantization codebooks. Aaron E. Rosenberg and Frank K. Soong (Speech Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

Highly efficient and effective short-term spectral representations of talkers can be obtained using vector quantization (VQ) codebook construction techniques. A talker recognition system has been implemented in which each talker is represented by a VQ codebook constructed from a large set of short-term spectral vectors obtained from a series of training utterances provided by the talker. In operation, the utterances of an unknown talker are analyzed and "encoded" using the codebook of a specified talker. The accumulated distortion between the input utterances and the specified talker's codebook is used to carry out a talker recognition decision. This technique can be said to be text independent to the extent that the training utterances adequately represent each talker's speech sound repertoire. The system can be extended to text-dependent operation with an additional training procedure in which specified utterances provided by a given talker are represented as encoded vector sequences using the talker's codebook. In use, an unknown talker is prompted to provide specified utterances which are analyzed and compared with the encoded prototypes for a specified talker. The system has been evaluated using a 100-talker database of 20 000 digits spoken in isolation. In a talker verification mode, average equal-error rate performance of 2.2% for text-independent operation and 0.3% for text-dependent operation is obtained for seven-digit-long test utterances.

Contributed Papers

3:30

Q5. Vector predictive quantization of the LPC spectral parameters for low-rate speech coding. Yair Shoham (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Scalar quantization of the LPC parameters requires a high bit rate. Considerably lower rate can be obtained via vector quantization (VQ). However, complexity constraints dictate the use of suboptimal VQ. The use of vector predictive quantization (VPQ) for the LPC parameters is proposed. In VPQ, the suboptimality is compensated for by exploiting the temporal redundancy in the input. VPQ is a two-stage memory VQ. In the first stage, the input vector is predicted from quantized past vectors, using a set of vector coefficients, held in a *predictor codebook*. In the second stage, the predicted vector is combined with a residual vector to form the final output. A set of residual vectors is held in a *residual codebook*. VPQ was applied to the LPC parameters in the down-sampled, log-magnitude spectral domain. The idea was to design an efficient VPQ under the perceptually meaningful log-likelihood distortion measure, while circumventing the stability problem of the synthesis LPC filter. This VPQ was used in a 4.8 kbit/s CELP coder where only 1.0 kbit/s were allocated to the parameters. The performance was almost indistinguishable from that of a CELP coder with unquantized parameters. [Work supported by NSA.]

3:45

Q6. Waveform speech coding based on vector transform quantization. Vladimir Cuperman (Tel-Aviv University and Calltalk Ltd., P. O. Box 45, Rishon Le Zion, Israel)

This paper presents a waveform speech coding system based on vector transform quantization (VTQ). VTQ is a coding system where consecutive M samples of a waveform are transformed into a set of coefficients which are quantized by a set of $m \ll M$ vector quantizers (VQs). A coding gain higher than for the known scalar transform coding systems is achieved by employing vector quantization rather than the usual scalar quantization. Moreover, a bit allocation closer to the optimal one may be achieved by vector quantizers as a result of improved round-off conditions. Assuming stationary input, the optimal bit allocation for the vector case is derived, and the coding gain achieved by such a system is evaluated. Techniques for adapting the bit assignment to the changing speech statistics are presented. These techniques are specific for the vector quantization approach and achieve adaptation along time and frequency axes. System simulation results are presented for typical speech waveforms at rates of 1–2 bits/sample.

4:00

Q7. A VQ-based speaker normalization method and its applications to speech recognition. Frank K. Soong (Speech Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

A speaker-specific vector quantization (VQ) codebook was proposed and applied successfully to both text-independent and text-dependent speaker recognition applications [e.g., F. K. Soong, A. E. Rosenberg, L. R. Rabiner, and B-H. Juang, ICASSP-85 (1985) and A. E. Rosenberg and F. K. Soong, ICASSP-86 (1986)]. In this talk the same VQ codebook is used for constructing a transformation, or more precisely, a mapping, between the feature space of a new speaker and that of a standard speaker. The mapping is constructed by using standard dynamic programming (DP) procedure to align training tokens spoken by a new speaker with tokens of the same text spoken by the standard speaker. Along the optimal alignment paths, a correspondence between the spectral feature vectors of the new speaker and the VQ codebook indices of the standard speaker is established and, for each VQ codebook index, a centroid is computed as the "average" of all the corresponding feature vectors of the new speaker. A new VQ codebook for the new speaker is therefore generated, and it is a one-to-one projected image of the VQ codebook of the standard speaker onto the feature space of the new speaker. This speaker normalization procedure has various applications in speech signal processing such as speech recognition, speech coding, and text-to-speech synthesis, etc. In this talk, isolated word recognition results are used as a demonstration of this new speaker normalization procedure. Results obtained from using speaker-adapted templates (the new method), speaker-independent clustered templates, and speaker-trained templates are compared.

4:15

Q8. Stochastic vector quantization of LPC parameters. B. S. Atal (Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

In this paper various issues related to efficient low-bit-rate coding of LPC parameters are discussed. It has been shown previously [Juang *et al.*, IEEE Trans. Acoust. Speech Signal Process. ASSP-30, 294–304 (1982)] that vector quantization provides a significant reduction in bit rate for coding the LPC parameters. Vector quantizers have two main disadvantages: To keep the quantization distortion to a low level, a very large codebook of vectors is usually needed. It is very difficult to train such large codebooks on real speech data and it is impractical to search these codebooks for optimum codewords in real time. In this paper, a vector quantization procedure which uses a codebook of white Gaussian random numbers to encode the LPC parameters is discussed. Interparameter correlations of LPC parameters are estimated from previously quantized data and are used to create codewords with the correct distribution. The optimum codeword is selected by an exhaustive search to minimize a weighted Euclidean distance between original and quantized parameters. Use of random codewords eliminates training of the codebook and provides robust performance over different speakers and speaking environments.

4:30

Q9. A fully quantized stochastic coder for low-bit-rate speech coding. Peter Kroon (AT&T Bell Laboratories, Murray Hill, NJ 07974)

Stochastic coding has the potential of providing high-quality speech at low bit rates [Schroeder and Atal, ICASSP (1985), pp. 937–940]. In this scheme linear predictive techniques are used to model both the long-term and short-term correlations in the speech signal. The remaining residual signal (excitation) is modeled by a codebook populated with samples from a Gaussian source. The whole procedure is based on an analysis-by-synthesis approach that minimizes the error between the original and the reconstructed signal. In the coder described by Schroeder and Atal, the excitation was coded at 2 kbps and the remaining parameters were left unquantized. To assess the usefulness of the stochastic coding approach for low-bit-rate speech coding, the performance of the coder with all the parameters quantized was investigated. A scalar quantizer was used to encode the short-term predictor coefficients at 1.8 kbps. Different vector quantization procedures for the long-term predictor coefficients resulting in bit rates varying from 1 to 2 kbps were investigated. It was found that a total of 8 kbps were required to produce results undistinguishable from the unquantized version. Speech coded at different rates between 4.8 and 8 kbps will be played at the conference. [Work supported by NSA.]

4:45

Q10. On comparing tree and codebook coding of LPC residuals at very low bit rates. Daniel Lin (Bell Communications Research, Inc., 435 South Street, Morristown, NJ 07960)

It has been suggested that codebook coding of LPC residuals offers significant performance advantage over tree coding at very low rates [cf. M. R. Schroeder and B. S. Atal, Proc. ICASSP 85]. An innovation codebook provides a better covering of the LPC innovation space (in the sense of having a smaller covering radius) than an innovation tree of the corresponding rate. These comparisons usually assume that the branches of the innovation tree are populated with independent identically distributed random variates (i.e., a stochastic tree). In this paper, the methods of selecting the innovation elements on a binary innovation tree such that the branches contain "complementary" sequences are examined. The covering properties of these "complementary" tree codes are compared with the covering properties of stochastic trees and stochastic codebooks. The results on the "complementary" tree codes indicate that (1) the performance of an innovation code did not correlate with its covering radius and (2) tree coding of LPC residuals at very low bit rate (rate 1/4 code) did not result in any performance degradation.

TUESDAY AFTERNOON, 9 DECEMBER 1986 — SALONS G AND H, 2:00 TO 4:45 P.M.

Session R. Underwater Acoustics II: Quality Assessment of Numerical Codes, Part 2: Benchmarks

Leopold B. Felsen, Chairman

Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, New York 11735

Invited Paper

2:00

R1. Benchmarks: Are they helpful, diversionary, or irrelevant? Leopold B. Felsen (Polytechnic University, Farmingdale, NY 11735)

There is universal agreement that computer codes should be accurate, but less consensus as to how they should be tested. This talk summarizes the outcome of a special session on Benchmarks held at the First IMACS Symposium on Computational Acoustics, Yale University, 6–8 August 1986. Attention is then given to analytical benchmark solutions, with emphasis on idealized but exactly solvable range-dependent coordinate separable and nonseparable models. After describing what information can be extracted from such benchmarks, the question of their utility for testing codes dealing with "real world" situations remains. This discussion should set the stage for the panel discussion that follows. [Work supported by ONR.]

Contributed Papers

2:25

R2. The multilayer expansion as a benchmark for range-independent propagation loss models. Henry Weinberg (ODSI Defense Systems, Inc., North Stonington Professional Center, North Stonington, CT 06359-1738)

In today's world of normal mode, FFP, and parabolic equation codes, one seldom thinks of ray tracing as a benchmark for propagation loss models. However, the former are often inappropriate in bottom-bounce regions, and do not readily provide physically significant quantities such as arrival angle and travel time. The multilayer expansion of the reduced wave equation is a theoretically exact ray model for range-independent environments. The numerical implementation approximates the ocean sound-speed profile by layers in which inverse sound speed squared is linear and the density is constant. The depth-dependent Green's function is then expanded into a finite geometriclike series plus a remainder. In many applications the remainder is small and may be neglected. For benchmark purposes, one must integrate the remainder over wavenumbers of interest in order to obtain the total field. [Work supported by the Naval Research Laboratory.]

2:40

R3. Modifying normal mode models to permit special range dependence. Dong-Jye Li (University of Rhode Island, Kingston, RI 02881) and David H. Wood (University of Rhode Island, Kingston, RI 02881 and Code 3332, Naval Underwater Systems Center, New London, CT 06320)

Benchmarks for testing range-dependent computer models are very scarce. We suggest that well validated normal mode models having no range dependence can be modified to permit their modeling a range dependence of a special form having parallel boundaries. This would permit modeling a three-dimensional index of refraction of the special form $n^2 = a(z) + c + b/(x^2 + y^2)$, where $a(z)$ is any depth-dependent function that an existing normal model will accept. Existing normal mode models are found to need only minor coding changes to permit generating special three-dimensional solutions of the Helmholtz equation or solutions of the standard parabolic equation.

2:55

R4. An exact modal solution in two dimensions. John A. DeSanto (Center for Wave Phenomena, Mathematics Department, Colorado School of Mines, Golden, CO 80401)

For a one-dimensional sound-speed profile varying in depth, it is known that there is an exact integral transformation relating the solutions of the Helmholtz and parabolic equations. This relationship can be extended to two dimensions using conformal mapping techniques. In particular a class of range- and depth-dependent sound-speed profiles can be generated for which (a) the Helmholtz and parabolic solutions are related by an exact integral transform, (b) both have an exact model expansion, and (c) asymptotically the Helmholtz and parabolic modal amplitudes and phases can be explicitly compared to show their differences. Constants can be chosen to preserve one mode amplitude and one mode phase. Profile examples are presented. The only necessary computational capability is a normal mode code but here used in mapped coordinates.

3:10

R5. An exact solution to the ideal, 3-D wedge as a proposed benchmark. M. Buckingham (Massachusetts Institute of Technology, Cambridge, MA 02139 and Royal Aircraft Establishment, Farnborough, Hampshire GU14 6TD, England), A. Tolstoy, and R. Doolittle (Code 5120, Naval Research Laboratory, Washington, DC 20375)

An exact solution for the field from a cw point source in an ideal wedge exists in the form of normal modes where the mode coefficients contain the entire range and cross-range dependence. This model is proposed as a benchmark 3-D solution. For the case when the wedge angle is a submultiple of π the solution has been implemented on a VAX 11/750, and results are available for the following: (a) the complete complex field throughout the wedge including the vicinity of the source; (b) individual mode coefficients (amplitude and phase); (c) transmission loss as a function of position; and (d) the spatial coherence of the field. The model is suitable for a general sensitivity analysis.

3:25

R6. The use of replica correlation to benchmark acoustical propagation models. B. B. Strozeski (ODSI Defense Systems, Inc., North Stonington Professional Center, North Stonington, CT 06359-1738), P. D. Herstein (Naval Underwater Systems Center, New London, CT 06320), and H. Weinberg (ODSI Defense Systems, Inc., North Stonington Professional Center, North Stonington, CT 06359-1738)

The three usual techniques employed for benchmarking acoustical propagation models involve measured acoustical data, analytical solutions, and other established acoustical propagation models. Each of these has inherent limitations. The alternative approach described here involves replica correlation to test the accuracy of acoustical propagation models. A received acoustic pressure spectrum, which is generated by the acoustical propagation model to be tested, is multiplied with the source pressure spectrum. The inverse Fourier transform of this product is a replica correlogram, which contains information of the travel time, phase and amplitude of all arrivals. Then a ray-tracing model is used to determine the travel time of the arrival. If the model to be tested is accurate, the travel times computed by replica correlation will agree with the travel time computed by ray tracing. Results will be presented for an existing acoustical propagation model under realistic ocean conditions. [Work supported by NRL.]

3:40

R7. Benchmark problems for broadband cross-correlation models in a nonisospeed medium. W. Hauck (Naval Underwater Systems Center, New London, CT 06320), P. Bilazarian (Raytheon Company, 1847 W. Main Road, Portsmouth, RI 02871), and P. Herstein (Naval Underwater Systems Center, New London, CT 06320)

Some benchmark problems for broadband cross-correlation models in a nonisospeed medium are proposed. Several depth-dependent profiles are employed, which have either analytic expressions or very accurate numerical solutions for significant ray acoustic quantities. These profiles include: (1) an isovelocity layer above an isogradient layer, and (2) bilinear profiles for both bounded and unbounded media. Several multipath propagation conditions, including direct path, bottom bounce, and convergence zone, are discussed. For these benchmark problems, the spectral contents of similar eigenrays are assumed to be equal; however, the power between any two paths may vary by a scale factor. By assuming a specific source–receiver geometry and input signal spectrum, the cross-correlation function can be expressed in terms of ray travel time and amplitude differences and signal bandwidth. Thus the accurate computation of eigenray quantities can be used to evaluate broadband cross-correlation models.

Panel Discussion: What Benchmarks are Relevant?

Panel Members

L. B. Felsen, Moderator
Polytechnic University, Farmingdale, New York 11735

D. Lee
Naval Underwater Systems Center, New London, Connecticut 06320

D. H. Wood
Code 3332, Naval Underwater Systems Center, New London, Connecticut 06320

F. B. Jensen
SACLANT ASW Research Center, 19026 La Spezia, Italy

R. A. Stephen
Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543

J. A. De Santo
Center for Wave Phenomena, Mathematics Department, Colorado School of Mines, Golden, Colorado 80401

Session S. Architectural Acoustics III: Acoustical Evaluation of Halls for the Performing Arts, Part 3

Ewart A. Wetherill, Chairman

Wilson Ihrig and Associates, 5776 Broadway, Oakland, California 94618

Contributed Papers

9:00

S1. Integration of acoustics and architecture in the auditorium at the Metro Toronto Convention Center. Ray Van den Broeck (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

The new Metro Toronto Convention Center Hall is a one-balcony auditorium having a seating capacity of 1350. The space is designed as a stage house multipurpose auditorium. Excellent acoustical environments are achieved for functions such as: lectures, motion pictures, speech, and music drama and music recital. The paper describes how final architectural forms and materials of the stage and the house, seating configuration and type, as well as surface size, shaping, and orientation were influenced by row-to-row sightlines, distance between the audience and stage, eccentricity angle, vertical viewing angle, under balcony isolation, safety, circulation, and the acoustical requirements: clarity/intelligibility, balanced and uniform sound projection, cohesion for performers on stage, freedom of echoes, strong envelopmental sound, reverberant sound level, and decay rate.

9:15

S2. Acoustical evaluation of halls for the performing arts using acoustical models. Bertram Y. Kinzey, Jr. and Gary W. Siebein (Department of Architecture, 231 ARC, University of Florida, Gainesville, FL 32611)

A series of relatively new objective measures of the acoustical quality of performing arts halls such as temporal energy ratios, early decay times, and rise times has been investigated and compared to each other and the perceived quality of the halls by many researchers. Measurements of these criteria were made at multiple positions in 1:10, 1:40, and 1:100 scale models of a 1500-seat multipurpose performance hall and in equivalent positions in the prototype room. There were clearly distinguishable zones of differing acoustical response shown in reflectograms made in the prototype room and even the smallest models. The objective measurements taken in the small size models with wideband noise duplicated the measurements taken in the larger size models and the prototype room reasonably well. The measurements made on an octave-band basis in the 1:10 scale model and the prototype room at equivalent positions show acceptable correlation. Twenty thousand hertz was the midband frequency for the highest octave band in the models that yielded acceptable values without correcting for air absorption. The ease of testing and the comparative results make this a valuable tool for preliminary testing of design alternatives.

9:30

S3. Measurement of panel reflection using acoustical scale modeling techniques—Part I: Instrumentation and procedures. Jose C. Ortega (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

It has been recognized for some time that the size, shape, and orientation of reflecting panels used in enclosed spaces affect their reflecting characteristics. This paper describes the test chamber, instrumentation, and procedures that have been developed to measure reflecting characteristics of scale model panels.

9:45

S4. Measurement of panel reflection using acoustical scale modeling techniques—Part II: Measured results. Jose C. Ortega and Paul S. Veneklasen (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

The importance of early reflected sound from side wall and ceiling reflecting panels in auditoria and other spaces used for musical presentation has been recognized for some 25 years. Quantitative data on the size, shape, and orientation of reflecting panels to achieve wideband spectral reflections have not been widely available. This paper presents some results on the effect of size and aspect ratio on the reflection characteristics of hard panels measured using scale modeling techniques.

10:00

S5. *In situ* auditorium measurements of quadratic residue diffusors. Kurt M. Graffy and Dennis A. Paoletti (Paoletti Lewitz Associates, Inc., 40 Gold Street, San Francisco, CA 94133)

A recently completed music hall in Moscow, Idaho has incorporated as a major portion of its renovation, quadratic residue diffusors (QRDs). The QRD surfaces were sensitively incorporated by the architect into the 1935 "collegiate gothic architecture" of the hall. In addition to subjective evaluations, objective measurements have been made to determine the field behavior of the QRDs in the vicinity of the performers' environment. Time/energy/frequency measurements and polar scattering plots have been made and compared to computer calculations and plots of the same. The hall is the 750 seat Administrative Auditorium Renovation at the University of Idaho. The Architects were Hummel, Jones, Hunsucker, Miller; with Nelson Miller as the Principal in Charge and Project Architect.

10:15

S6. New room acoustics measurement software. J. S. Bradley and R. E. Halliwell (Institute for Research in Construction, National Research Council Canada, Ottawa, Ontario K1A 0R6, Canada)

Auditorium acoustics research has now developed to the point where there is quite general agreement on the subjective importance of four types of newer measures. These are: (1) decay times; (2) early/late ratios and other measures of clarity; (3) the overall strength or sound level; and (4) lateral energy fractions. In the past such measurements have been made by researchers using various complicated arrangements. The new software runs on an IBM PC compatible computer and controls a Norwegian Electronics two-channel analyzer, so that these newer auditorium acoustical measurements can be much more conveniently obtained. The design of the system and results by this and other methods will be discussed.

10:30

S7. The noncorrelated sound-pressure level: A quantifiable metric for spatial characteristics in performance spaces. Christopher N. Blair (Music Department, Massachusetts Institute of Technology, Cambridge, MA 02139)

Recent research at MIT's Experimental Music Studio in binaural signal processing has led to a simple technique for separating the correlated portion of an audio signal received by a listener (the quasimonophonic portion related to the direct sound path, nondiffuse first-order ceiling reflections, etc.) from that portion of the signal in which the amplitude and/or phase relationships differ at each ear (due to lateral reflections of different path length, randomly arriving diffuse reflections, etc.). The noncorrelated sound-pressure level is easily measured by comparison of the *relative* voltage outputs of the microphones in the binaural head using a sound level meter. Real heads exhibit unsystematic amplitude and phase anomalies which render perfect cancellation of monophonic signals impossible. However, for the unequalized head used in this study, the noncorrelated sound-pressure level of monophonically presented pink noise is more than 10 dB below the sound-pressure level measured in octave bands by each ear separately. Future work and applications are discussed. [Work supported by MIT Provost Fund.]

10:45

S8. Time-delay spectrometry in auditoria. Ronald L. McKay (BBN Laboratories, Inc., 21120 Vanowen Street, Canoga Park, CA 91303)

A variety of measurements using a time-delay spectrometer in European and American concert halls and multipurpose auditoria will be presented and interpreted. Direct/reverberant energy ratios, reverberation times, lateral energy ratios, orchestra platform effects, and varying sound reflection patterns will be covered.

11:00

S9. "Transition loss": From the practice room to the concert hall. Arthur F. Niemoeller and Punita Singh (Central Institute for the Deaf, 818 S. Euclid, Saint Louis, MO 63110)

The acoustical characteristics of music practice facilities are compared with those of some performance halls available at a university campus. Glaring quantifiable differences are observed between the practice spaces and performance spaces in terms of size, shape, reverberation, and interior construction schemes. Furthermore, the perceived acoustical qualities of three performance spaces studied also vary greatly from each other, one being popularly referred to as "very reverberant and muddy," another "dead," and a third, "good." The physical differences observed are presented, and their effect on the performance of music rehearsed in the small practice facilities and performed in the larger auditoria considered via subjective reports made by performers and listeners. These observations are further supplemented by an account of strategies employed by performers in adapting production to suit the different spaces, to compensate for the "transition loss" incurred in going from the practice room to the concert hall.

11:15

A tour of the new Orange County Performing Arts Center will be available following the conclusion of Session S. A bus will leave the hotel about 11:45 a.m. and return about 1:15 p.m.

WEDNESDAY MORNING, 10 DECEMBER 1986 — SALON 3, 9:00 TO 11:20 A.M.

Session T. Biological Response to Vibration III: Measurement of Tactile Sensation

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

Chairman's Introduction—9:00

Invited Papers

9:05

T1. Development of vibrotactile measurement techniques for assessing mechanoreceptor performance at the fingertip. J. E. Piercy and A. J. Brammer (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

In recent years knowledge of the basic physiology of mechanoreceptors has advanced considerably. The translation of this knowledge into psychophysical threshold measurement techniques for assessing mechanoreceptor performance at the fingertip will be described. The sensitivity of three types of mechanoreceptors, the Pacinian corpuscles, Meissner corpuscles, and Merkel disks, is obtained from threshold measurements with sinusoidal vibratory stimuli at various frequencies across the band from 2–400 Hz. Parameters affecting the coupling of the vibrating probe to the finger, namely, static force, skin indentation, and probe diameter, have a significant and different effect on the thresholds for the different receptors. The role of background vibration of physiological origin (physiological noise) on the perception thresholds will be discussed.

9:35

T2. Measurement of tactile spatio-temporal sensitivity. Clayton L. Van Doren (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244)

The perception and discrimination of texture is an important part of cutaneous tactile function, and requires the use of both spatial and temporal information. Earlier studies that have used a variety of stimulators to measure tactile responses to temporal and spatial variations will be reviewed. A stimulus that has been successfully employed in the study of visual pattern perception, but has not been available for tactile research, is the spatio-temporal sinusoid, where the amplitude of the stimulus varies sinusoidally along one spatial dimension and in time. A new, 88-element linear stimulator array, constructed to produce tangible spatio-temporal sinusoids, is described. Each element in the array is a plate of a piezoelectric ceramic. The elements are assembled on 0.38-mm centers for a total array length of 33.4 mm. Therefore, the spatial wavelengths that can be produced by the array range from 0.76–33.4 mm. The temporal bandwidth of the array is 0.8–1000 Hz, and the maximum stimulus amplitude is 8 μ zero to peak. The array was used to measure detection thresholds at five temporal frequencies (1, 4, 16, 64, and 256 Hz) and five spatial wavelengths (1.81, 3.62, 7.23, 14.5, and infinite mm). The detection thresholds reveal two receptor populations, presumably the NP I and P systems described by Verrillo and colleagues. The NP I system is more sensitive than the P system at short wavelengths and at low temporal frequencies, and less sensitive than the P system at high frequencies and long wavelengths. These results are consistent with earlier psychophysical and physiological experiments.

10:05

T3. Detection of whole-nerve sensory action potentials generated by vibrotactile stimulation of a fingertip. A. J. Brammer (Division of Physics, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada), I. Pyykkö, P. Kolari, and J. Starck (Departments of Physiology, and Industrial Hygiene and Toxicology, Institute of Occupational Health, Helsinki, Finland)

Current electrophysiological techniques for studying whole-nerve properties in man employ electrical stimulation of nerve pathways. They thus cannot provide information on the cutaneous nerve endings. This limitation may be overcome for the touch receptors by vibrotactile stimulation. The resulting electrical activity is recorded by wire electrodes looped around the finger, and surface electrodes located above the nerve at the palm and wrist. After signal processing to reduce physiological noise, a pulse train is recovered with peak to peak amplitude of, typically, 0.2 μV. With short duration (1.0 ms), approximately half-sine-wave pulse stimulation, and resolution imposed by the residual noise ($\sim$0.05 μV peak-peak), the sensory action potentials consist of from one to two cycles of a near 1000-Hz sinusoidal pulse. Action potentials have been detected at, and above, 20-dB sensation level, corresponding to skin displacements of the order of 20 μm. With sinusoidal stimulation, signals believed to be action potentials have been recorded only from selected subjects. The application of this technique to the assessment of mechanoreceptor and sensory nerve function will be discussed.

WEDNESDAY AM

Contributed Papers

10:35

T4. Comparison of two single-channel vibrotactile aids for the hearing impaired. Janet M. Weisenberger and Ann F. Russell (Central Institute for the Deaf, 818 S. Euclid, Saint Louis, MO 63110)

Two commercially available single-channel vibrotactile aids for the hearing impaired were compared in tasks using acoustic stimuli of varying levels of complexity. Both devices provide envelope information in an AM signal presented to a single location on the skin; however, one of the devices varies the carrier frequency according to input frequency, providing some additional "spectral" information about the stimulus. Subjects were tested in tasks including sound detection, environmental sound identification, syllable number and stress identification, and phoneme identification of limited-set vowels and consonants. All stimuli were recorded onto disk and presented through a loudspeaker by computer. Results showed rapid learning of tasks requiring only envelope information (environmental sounds, syllable number and stress), but little or no improvement with training in vowel and consonant identification. No significant differences between the devices were found on any task, suggesting that the additional spectral information provided by one of the devices is not effectively utilized by the tactile system in the absence of visual cues. [Work supported by NSF and NIH.]

10:50

T5. The effect of a wearable 16-channel electrotactile sensory aid on receptive communication skills of profoundly deaf children. Barbara Franklin (Department of Special Education, San Francisco State University, San Francisco, CA 94132) and Frank A. Saunders (Smith-Kettlewell Institute of Visual Sciences, San Francisco, CA 94904)

Six children, 3 to 8 years of age, with profound congenital binaural sensorineural hearing losses were fitted with a wearable elecrotactile sensory aid, the Tacticon, in March 1985. Their receptive skills for both speech and environmental sounds were evaluated using a battery of selected tests. Following a 9-month training period, a comparison was made of their scores in the following conditions: (1) pre- and post-test auditory only (with hearing aids); (2) pre- and post-test auditory/visual (hearing aids and speechreading); and (3) post auditory/tactile (hearing aids and Tacticon) and auditory/visual/tactile (hearing aids, speechreading, and Tacticon). All the children were able to integrate information delivered simultaneously by the auditory, visual, and tactile senses and their scores were consistently higher when wearing the Tacticon than in the comparable condition without the tactile aid. A modified "tracking" procedure was used to determine the amount of time required to correctly identify spoken utterances. There was a substantial reduction in time in the post-test auditory/visual/tactile condition compared to the pre-test auditory/visual condition. [Work supported by NIH.]

11:05

T6. Acoustical behavior of ringed seals in the presence of man-made noise. William C. Cummings (Oceanographic Consultants, 5948 Eton Court, San Diego, CA 92122) and D. V. Holliday (Tracor, Inc., 9150 Chesapeake Drive, San Diego, CA 92123)

A total of 24 373 underwater sounds were recorded from ringed seals (*Phoca hispida*) in Kotzebue Sound, AK. Of these, 81% were from 40–500-ms seal scratches used in lair and access or breathing hole maintenance (1–6 kHz, 98–102 dB *re*: 1 μPa at 1 m). Remaining seal sounds were vocalizations consisting mostly of rublike sounds, squeaks, and quacking barks (80 ms to 1.5 s, 0.5–9 kHz, 95–131 dB *re*: 1 μPa at 1 m) that peaked in occurrence at 1930 h and were more frequent during lightest (450–570 $\mu W/cm^2$) hours. Vocalization was negatively correlated with windspeed and ambient temperature; scratch sounds were not correlated with temperature. Our introduction of underwater man-made noise (playbacks of on-ice Vibroseis® operations and control noises) had no statistically significant effect on sound production (three playback sessions of 14.5 h, compared to 3000 vocalizations, with control periods before, during, and after playback), but an underlying long-term increase (6 $\times$) existed, presumably from heightened breeding and parenting activity. Man-made low-frequency pulses of unknown origin illicited immediate vocal outbursts. [Work supported by NOAA.]

WEDNESDAY MORNING, 10 DECEMBER 1986 SALONS 4 AND 5, 9:00 A.M. TO 12:15 P.M.

Session U. Noise IV, Shock and Vibration IV, and Musical Acoustics III: FFT Analyzers for Noise and Vibration from a User's Perspective

Frank H. Brittain, Chairman
Bechtel National, Inc., 50 Beale Street, San Francisco, California 94105

Chairman's Introduction—9:00

Invited Papers

9:05

U1. Single-channel FFT analyzers. Frank H. Brittain (Bechtel National, Inc., 50 Beale Street, San Francisco, CA 94105)

FFT analysis gives engineers powerful tools to diagnose and solve problems, and the power to get into trouble. As the sophistication and number of buttons increase, the difficulties in getting valid and useful results also increase. Fundamentals of single-channel FFT analyzers and their capabilities are discussed to help the user better understand the practical use of FFT analyzers. The concept of time and frequency domains is developed using an analog model. Assumptions implicit in using the finite Fourier transform to approximate the (infinite) Fourier transform are defined. Sampling rate, frequency range, number of lines, resolution, and sampling period are reviewed. Folding or aliasing and how to avoid it are illustrated; leakage and the use of windows to reduce errors are discussed. Various types of averaging are reviewed. A description is provided of how these fundamentals, inherent in all FFT analyzers, appear to the user of a single-channel analyzer. Provisions for mitigating errors built into most analyzers are outlined. The two basic outputs (frequency or auto spectra) and special output formats of typical single-channel analyzers are demonstrated.

9:25

U2. Dual-channel FFT analyzers. David A. Kienholz (CSA Engineering, 560 San Antonio Road, Suite 101, Palo Alto, CA 94306)

Adding a second channel to a Fourier analyzer greatly increases its capability by allowing investigation of the relationship between two signals which occur simultaneously. A number of important two-channel frequency functions can be measured, all derived from the cross-power spectral function. These include frequency response, ordinary coherence, and sound intensity. This presentation reviews the definitions, physical significance, and some practical applications of these data types. Practical considerations for two-channel measurements with typical dual- or multiple-channel analyzers are discussed.

9:45

U3. The FFT—From Fourier transform to application. H. J. Weaver (Lawrence Livermore Laboratory, 7000 East Avenue, Mail Stop L-194, Livermore, CA 94550)

This paper will present a discussion as to how the Fourier transform and/or frequency response function generated by spectrum analyzers can be used to gain insight and an understanding of the dynamic properties of noise and/or vibrational problems. A single degree of freedom (spring–mass–damper) system will be used as the focal point of this discussion. We will present a heuristic discussion of the frequency response function of this single degree of freedom system and show how its dynamic properties can be directly related to the graphical characteristics of the frequency response function. We will also examine how the coherence function can be used as a measure of the "goodness" of the frequency response function.

10:05

U4. Problems inherent in the utilization of the FFT. Randall J. Allemang (University of Cincinnati, Department of Mechanical Engineering, Rhodes Hall ML 72, Cincinnati, OH 45221)

The effective utilization of the FFT depends upon the understanding of the problems involved in implementing the FFT. These problems result from errors associated with three aspects of the implementation: errors associated with the time truncation of the data, errors associated with the acquisition of the digital data, and errors associated with the FFT algorithm. These errors can be categorized according to the effect of the error on resultant data. With respect to estimation concepts, these errors can be categorized as random or bias errors. Typical well-known bias errors are aliasing and leakage. The physical source and severity of common random and bias errors will be presented. The effect of the resultant data can be minimized through the use of different digitization hardware, different digitization parameters, different FFT transform size, frequency-shifted FFT concepts, averaging, weighting, and signal choice. Examples of such common error reduction methods will also be presented.

10:25

U5. Getting valid data. William J. Atherton (Cleveland State University, Department of Mechanical Engineering, 1983 East 24th Street, Cleveland, OH 44115)

The advent of microprocessors has made modern high-speed spectral analyzers possible and has given the test engineer new and powerful techniques for the analysis of structures. However, with new techniques have come new difficulties to overcome in obtaining valid data. This paper stresses the input/output nature of dynamic system measurements and builds upon this concept for the determination of valid data. First, consideration is given to various excitation techniques (swept sine, random noise, impulse) in addition to self-excitation. The distribution of energy in the frequency domain of the excitation signal offers insight as to what may be expected in the output measurement. Undesirable energy in a selected frequency range may need to be filtered out to better match the dynamic range of a signal with the input amplifier gain. Noise, which typically occurs on every signal, affects measurement accuracy. Averaging techniques are described which can sometimes help reduce the effects of noise and improve the measurements. In two-channel measurements, the use of the coherence function as a measure of the goodness of frequency response data is also introduced.

10:45

U6. Practical solutions of engineering problems through the application of FFT analysis. Martin W. Trethewey (Department of Mechanical Engineering, The Pennsylvania State University, University Park, PA 16802)

The analysis of many actual engineering problems can be greatly enhanced through the application of fast Fourier transform (FFT) signal processing. The purpose of this paper is to demonstrate some typical problem solving approaches and subsequent solutions obtainable through FFT analysis. The problems and solution techniques are chosen to illustrate the use of the various available spectral quantities. The topics discussed include: (1) source identification and preventative maintenance monitoring through comparison of power spectra; (2) the estimation of radiated acoustic intensity using the cross spectrum between two closely spaced microphones; (3) machinery noise source identification via the ordinary coherence function; and (4) the measurement and estimation of dynamic structural characteristics from frequency response functions. Techniques to assess the spectral quality for each application are discussed to ensure that accurate results are obtained. Each topic area is presented in a pragmatic fashion to illustrate the use and interpretation of the spectra required for each application.

11:05–11:15
Break

11:15

U7. Using FFT analyzers in undergraduate physics courses. Thomas D. Rossing (Department of Physics, Northern Illinois University, De Kalb, IL 60115)

FFT analyzers are very useful, not only in acoustics demonstration and laboratory experiments, but in courses in electronics, electricity and magnetism, mechanics, optics, etc. A few of the pedagogical applications of these useful instruments will be described.

11:30

U8. Windows in spectral analysis—what they do. John C. Burgess (Department of Mechanical Engineering, University of Hawaii, Honolulu, HI 98622)

"Windows" are functions used via multiplication/convolution to alter the spectral characteristics of signals represented by finite length data records. Since finite record lengths are always used for real-world spectral analysis, a window of some shape is always used in practice. The simplest window is the open window, the "hat" function by which an infinite Fourier integral becomes a finite Fourier integral. Windows of other shapes have two major functions that depend on the type of data. Windows are used with periodic data to minimize the effects of signal discontinuities at the "ends" of a data record. Windows are used with random data to improve statistical reliability of spectral estimates. Shapes intended to implement either application automatically affect the other. The assumption implicit in using the open window is that data outside the window are zero, while the assumption implicit in Fourier analysis is that the data record is periodically extended. These assumptions are often not valid.

11:45

U9. Recent experience with the new ANSI S12.10-1985 procedure for identifying prominent discrete tones. Larry E. Wittig and Robert D. Hellweg, Jr. (Products Acoustics Group, Digital Equipment Corporation, ML08-3/T13, 146 Main Street, Maynard, MA 01754-2571)

The American National Standard for measuring noise emitted by business equipment, ANSI S12.10-1985, includes a new procedure for identifying prominent discrete tones. The procedure is based upon comparing the sound pressure level in the tone to the sound pressure level in the Zwicker bandwidth centered around the tone. In general this new procedure is a substantial improvement over the previous procedure described in ANSI S1.29-1979, and based on Appendix B of ANSI S.13, which uses sound pressure levels outside the Zwicker bandwidth to determine the strength of the masking noise. However, after several months of use with a FFT analyzer it is clear that the S12.10 procedure raises questions that should be addressed. This paper presents examples of tones that sound prominent but do not calculate to be prominent by the new procedure, and examples that do not sound prominent but calculate to be prominent. This paper also presents limitations on use of one-twelfth-octave-band filters to make the measurements required by the S12.10 procedure.

12:00

U10. An interactive graphic system for acquiring and analyzing proportional bandwidth acoustical data. Matthew Sneddon and Sanford Fidell (BBN Laboratories, Inc., 21120 Vanowen Street, Canoga Park, CA 91304)

Computer-based systems for processing one-third-octave and other proportional bandwidth acoustical data are generally specific to particular applications (e.g., aircraft noise certification), narrow in range of data manipulation, inconvenient to extend to applications other than those for which they were designed, and require skilled operators. These limitations seemed tolerable when the cost of digital systems exceeded the price of human labor. Now that acoustical data analysis systems can be economically developed in high level languages, a different set of requirements for such systems is emerging. These include extensive interactive graphic analysis capability, advanced statistical treatments of data, ease of data management, user extendability to unforseen applications, generic procedures for data manipulation, simplicity of application to novel data sets, concurrent access to data sets by multiple analysts, and automated operation. An example of a modern, VAX-based acoustical data analysis system of this sort is described in detail.

WEDNESDAY MORNING, 10 DECEMBER 1986 — SALONS 1 AND 2, 9:00 A.M. TO 12:10 P.M.

Session V. Physical Acoustics III and Underwater Acoustics III: Special Session on Modern Topics in Acoustics in Honor of Isadore Rudnick's 70th Birthday

Steven L. Garrett, Chairman
Physics Department, Naval Postgraduate School, Monterey, California 93943

Herman Medwin, Co-Chairman
Physics Department, Naval Postgraduate School, Monterey, California 93943

Invited Papers

9:00

V1. First, a word from our sponsor. Logan E. Hargrove (Physics Division, Office of Naval Research, Arlington, VA 22217-5000)

The Office of Naval Research is pleased to have supported and to have been associated with Isadore Rudnick's unique research contributions in physical acoustics and wish him a happy 70th birthday.

9:05

V2. On shrinking the world's oceans into a physics laboratory. Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Traditional operations of a physical acoustics laboratory involve the use of sound as a probe to determine the physical constants and characteristics of fluids and solids. The same techniques can be used, and the same physical insight can be achieved, when one scales down complex extended oceanic phenomena to manageable laboratory dimensions. A scale model solution to underwater acoustics propagation problems has the advantage that the intimately intertwined character of the real world can be tested, one parameter at a time, in the laboratory micro-ocean. The unraveling of ocean acoustical mysteries has been performed to advantage in several problems: In the case of shadowing by a seamount, the specific contributions of diffraction at the seamount crest, multiple reflections between seamount and ocean surface, scatter at the rough bottom, and scatter from wind waves at the ocean surface have been clarified. Current research in our laboratory involves the evaluation of components of scatter from a model of the arctic ice canopy where reflection, specular scatter from steep-sloped roughness elements, boundary waves, diffraction at the edges, and flexural waves within the ice plate exist simultaneously. All these contributors play their parts in producing what has been innocently described as the "reflection coefficient" and the "backscattering strength" of the arctic ice cover. [Work supported by ONR.]

WEDNESDAY AM

9:25

V3. The long and short of random media: Acoustic Anderson localization. Julian D. Maynard (Department of Physics, The Pennsylvania State University, University Park, PA 16802)

Wave propagation in random media relates to a large number of important current problems in physics and acoustics. For long wavelengths the wave equation is locally approximated by the Laplace equation, and this together with the complex boundary conditions provided by the random medium describes many interesting phenomena involving percolation, fractional dimensionality, etc. An important puzzle in the long wavelength regime was solved just recently; a homework problem once assigned by I. Rudnick provided the key to the solution. In the short wavelength regime, where wavelengths are on the order of the variations in the random medium, one may observe effects of Anderson localization, which is currently receiving much attention in the study of quantum phenomena in disordered solids. Recent experiments on acoustic Anderson localization, including nonlinear effects, will be reported. [Work supported by ONR.]

9:55

V4. Nonlinear acoustics of a two-dimensional plasma in He^4. Seth Putterman, Bruce Denardo, and Gary A. Williams (Physics Department, UCLA, Los Angeles, CA 90024)

Balance between the repulsion of positive ions by a dielectric–vacuum interface and an externally imposed holding field facilitates the formation of a 2D plasma (net charge 10^8 e/cm^2) about 300 Å beneath the free surface of He^4 [Ott-Rowland, Kotsubo, Theobald, and Williams, Phys. Rev. Lett. **49**, 1708 (1982)]. Density modes of the charged sheet propagate at acoustic frequencies (10–100 kHz) and have a high adjustable reversible dispersion. The resonance frequencies were found to soften up to 30% at Mach numbers of 0.01. Continuum mechanics yields a good description of the linear dispersion law but to third order in amplitude theory predicts a frequency shift which is too small by a factor of 1000 and has the wrong sign. The huge nonlinearity makes this an ideal system to probe new solitons [Wu, Keolian, and Rudnick, Phys. Rev. Lett. **52**, 1421 (1984)] and may provide a means of probing Wigner crystallization via the use of an end fire array to achieve mode conversion [Garrett, Adams, Putterman, and Rudnick, Phys. Rev. Lett. **41**, 413 (1978)] of propagating 2D plasma oscillations to 2D shear modes.

10:20

V5. Surface acoustic wave interaction with thin magnetic films. Moises Levy and Roy Wiegert (Physics Department, University of Wisconsin–Milwaukee, Milwaukee, WI 53201)

It has been found that surface acoustic waves (SAW) exhibit a very large interaction with appropriately prepared thin magnetic films through the magnetoelastic effect. For a 600 Å 90Ni 10Fe thin film, the interaction can produce changes in attenuation of 30 dB/cm at 700 MHz by changing from 2 to 12 G a magnetic field applied parallel to the film plane and perpendicular to the SAW. Measurements of the frequency dependence of this large effect yield values for the Gilbert damping constant and the anisotropy field. This interaction has been studied in the series of xNi $(1-x)$Fe alloy films. For $x > 80$ wt %, the magnetoelastic constant η is negative. It is positive for $x < 80$ wt %. An amorphous film with positive η has also been studied. Some device concepts which make use of this effect will be presented. [Work supported by the National Science Foundation under Grant No. ESC 8519695.]

10:40

V6. Correlation between frequency and position of soliton creation. Brad Barber, Junru Wu, and Isadore Rudnick (Department of Physics, University of California, Los Angeles, CA 90024)

A water trough which has the property that there is a correlation between the frequency of drive and the position at which the soliton is created has been designed and constructed. It has a linearly increasing width and a quadratically increasing depth. This results in a trough which has a cutoff frequency for solitons which varies with position when driven parametrically. Self-trapping localizes the soliton to the same degree as in a conventional trough with constant rectangular cross section [J. Wu, R. Keolian, and I. Rudnick, J. Acoust. Soc. Am. Suppl. 1 **74**, S11 (1983)]. The observed correlation between frequency and position agrees unexpectedly well with an *ad hoc* approximate theory. [Work supported by ONR.]

10:55

V7. Nonlinear flexural modes in thin shells: Observation of envelope solitons. Junru Wu, John Wheatley,[a] and Isadore Rudnick (Department of Physics, University of California, Los Angeles, CA 90024)

Olson [AIAA J. 3, 1775 (1965)] measured the flexural eigenmodes of a thin metallic shell and found that the dispersion curve for the (p,q) mode, where p refers to azimuthal and q to height waves, has an unusual shape. When q is kept constant, the shape is that of a distorted U. We have reexamined this system and found it to be very rich in nonlinear properties. These phenomena include nonlinear mode interaction, multiple hysteresis, and quasiperiodicity. Of particular interest is the observation of envelope solitons propagating with a group velocity which is equal to the slope of the abovementioned dispersion curve. Since the sign of the slope is different in the two branches of the U, the solitons associated with each of them propagate in opposite directions. Observations of this behavior will be described. To our knowledge this is the first direct observation of solitons in macroscopic elastic waves in solids. [Work supported by ONR and DOE.] [a] Deceased.

11:10

V8. Measurements of the nonlinear tuning curves of thin circular disks. Junru Wu and Isadore Rudnick (Department of Physics, University of California, Los Angeles, CA 90024)

The flexural modes of a circular steel disk (diameter $\approx$25.7 cm, thickness $\approx$0.061 cm) were excited by an electromagnetic force to large amplitudes of motion, and the nonlinear tuning curves of the (p,q) modes, where p = nodal diameters and q = nodal circles, were determined. The results disagree with those of S. A. Tobias [Engineering **51** (11 July 1958) and Proc. Inst. Mech. Eng. London **171**, 691 (1957)]. It is found that, as the drive level is increased, the peak frequencies of the tuning curves can shift to lower frequencies, such as (0,1), (1,1), (0,2) modes, or higher frequencies, such as (10,0), (3,1), (4,0) modes; the direction of shifting is a property of the particular modes being driven. [Work supported by ONR and DOE.]

11:25

V9. Method for controlled rotation in an acoustic single mode levitator. J. L. Allen and M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, 4800 Oak Grove Drive, Pasadena, CA 91109)

A method for producing controlled rotation of an acoustically levitated object in a single mode cylindrical levitator has been developed and tested experimentally. In this method, two spatially separated acoustic drivers are used to provide the levitation and rotation capabilities. The levitation sound fields can produce two types of acoustic torques: a stabilizing "Rayleigh" torque and a rotating torque [M. Barmatz, J. Acoust. Soc. Am. Suppl. 1 **78**, 44 (1985)]. For an appropriately chosen mode, an amplitude variation and/or phase shift introduced between the two drivers can be used to unbalance the positive or negative going angular waves leading to a net rotational torque on the sample. A finite minimum unbalance of the angular waves is required to overcome the "Rayleigh" torque and produce rotation. The dependence of this minimum on the phase shift, chamber length, sound-pressure level, and sample size will be discussed and a videotape of the rotation experiments will be presented. [Work supported by NASA.]

11:40

V10. PVDF: A wideband ultrasound transducer at ultralow temperatures? Robert Keolian and John Reppy (Laboratory of Atomic and Solid State Physics, Cornell University, Ithaca, NY 14853)

The propagation of ultrasound in rotating superfluid He-3 requires that the power dissipated be limited to a nanowatt or so. The usual transducers at these millikelvin temperatures are quartz crystals driven at resonance. They have low loss but work at only a few fixed frequencies. The thin piezoelectric polymer film PVDF is sensitive enough that it can be used below its resonance, from audio to at least 60 MHz. Its suitability at ultralow temperatures will be described. [Work supported by NSF.]

11:55

V11. Resonant reciprocity calibration of conventional, magnetohydrodynamic, and quantumfluidic transducers. S. L. Garrett (Physics Department, Code 61 Gx, Naval Postgraduate School, Monterey, CA 93943)

Since its application to transducer calibration in 1940 by Cook and MacLean, the reciprocity method has been used for absolute calibration of electroacoustic transducers in a relatively limited number of geometries and media. Following Prof. Rudnick's article on extensions of this technique to unusual geometries [J. Acoust. Soc. Am. **63**, 1923 (1978)], the resonant reciprocity technique to high-precision (noncompliant) microphone and hydrophone calibration in air and water, to magnetohydrodynamic (ultracompliant) transducers in salt water and mercury, and to thermomechanical (second sound) transducers in superfluid ^{4}He has been applied. The technique for generalization of the theory to conducting and quantum fluids along with the experimental apparatus used to verify the generalizations will be described. [Work supported by the ONR.]

Session W. Psychological and Physiological Acoustics III: Auditory Evoked Potentials and Otoacoustic Emissions

David L. McPherson, Chairman
Department of Neurology, University of California, Irvine, California 92717

Contributed Papers

8:30

W1. Effects of relative starting phase and frequency separation on two-tone auditory brainstem responses. Carol Sammeth, Robert Burkard, and Kurt Hecox (Project Phoenix, Inc. and the Waisman Center on Mental Retardation and Human Development, 5225-4 Verona Road, Madison, WI 53711-0287)

Two-tone complexes have been used as stimuli in auditory brainstem response (ABR) studies examining two-tone suppression and the critical band (Stanny and Elfner, 1983; Zerlin, 1986). The relative starting phase of the tones in a two-tone complex can have dramatic effects on the stimulus envelope. In light of the onset nature of the brainstem response and its integration of energy over only the first few cycles of a tone-burst stimulus (Suzuki and Horiuchi, 1983), phase effects on the stimulus envelope may be a confounding factor in two-tone studies. We recorded ABRs to single-tone stimuli, and to two-tone complexes with homophasic and antiphasic relative starting phase. Frequency separations were arithmetically centered around 4000 Hz, and ranged from 200 to 3200 Hz. Relative starting phase had a significant effect on wave V amplitude for narrow frequency separations where stimulus envelope effects are most pronounced, but had no significant effect on wave V latency. ABRs to single-tone stimuli were summed and compared to the responses obtained with the two-tone complexes. The amplitude of the summed single-tone responses exceeded the amplitude of wave V of the two-tone complexes at all frequency separations. We conclude that relative starting phase of the tones in two-tone complexes can significantly influence wave V amplitude, especially for small frequency separations.

8:45

W2. Relationship of the auditory brainstem response to time/intensity tradeability. Mark Stephenson (Armstrong Aerospace Medical Research Laboratory, Biodynamics, and Bioengineering Division, Biological Acoustics Branch, Wright-Patterson AFB, OH 45433) and William Melnick (Department of Otolaryngology, The Ohio State University, Columbus, OH 43210)

Dichotic clicks were presented at 70 dB nHL to a group of 12 otologically normal male subjects. The clicks to one ear were attenuated by either 6 or 12 dB. Subjects were instructed to delay the onset of the click delivered to the contralateral ear until a single image was perceived at the center of their head. These same stimuli were then employed to evoke auditory brainstem responses. When the interaural intensity difference (IID) was 6 dB, a mean interaural time difference (ITD) of 151 μs effectively recentered the image. An IID of 12 dB was similarly offset by an ITD of 594 μs. A comparison of ABR latencies for time and intensity demonstrated greater differences for wave I than for wave II or V. The difference at wave I was significant for three out of four comparisons while at wave V only one out of four comparisons was significant. Differences between ABR response latencies to time versus those to intensity were most apparent as interaural differences were increased.

9:00

W3. Central masking paradigms and alterations in the auditory brainstem response. Radha Simhadri-Sumithra and George M. Gerken (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

Contralateral masking is often recommended in auditory brainstem response (ABR) testing in order to reduce or eliminate the participation of the nontest ear. The present study evaluates the effects of contralateral sound on the ABR. Vertex-mastoid recordings were made on a total of 30 young-adult subjects with normal hearing. Stimulus paradigms employed a 4-kHz probe and a contralateral continuous tone or a contralateral forward masker. Both the continuous-tone and forward-masking experiments resulted in statistically significant amplitude changes in the probe-evoked ABR. Waves III and VI, averaged across experiments, conditions, and subjects, showed a 28.7% increase in amplitude in the presence of a contralateral stimulus. In contrast, the amplitude of wave V, averaged across experiments, conditions, and subjects, was reduced by 11.2%. Latency measures for all waves often showed a slight, but statistically significant, increase. In regard to the enhancement of wave III by contralateral sound, we speculate that the descending auditory system is involved. The amplitude changes obtained for waves III, V, and VI contrast strongly with earlier results on ipsilateral masking.

9:15

W4. Effects of attention on auditorily evoked potentials. Francis K. Kuk and Paul J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Auditorily evoked responses to tone bursts (4-ms duration with 1-ms rise/fall) were recorded from the promontory as well as the ipsilateral mastoid of five human listeners while they were performing a visual duration (standard light flash of 50 ms) and an auditory frequency (center frequency at 2, 4, and 8 kHz; 250-ms duration) discrimination task. Response evoking tone bursts were presented prior to and between the discrimination tone bursts. Visual and auditory stimuli were presented simultaneously. ABR and AP measurements were made during separate sessions. The method of constant stimuli was used for the discrimination tasks and the separation between stimuli was set for the value of the difference limen determined during a training session. Recordings obtained with acceptable discrimination performance were analyzed. No difference in amplitude, latency, or shape of the recordings was observed between the two discrimination tasks. The results suggest that the attention demand of the present task is not a central factor in the control of the auditory efferents.

9:30

W5. Maturation effects on binaural interaction in human auditory middle latency evoked potentials. D. L. McPherson, C. Tures, and A. Starr (Department of Neurology, University of California, Irvine, Irvine, CA 92717)

An interaction of the binaural auditory pathway may be observed in human auditory evoked potentials by subtracting the evoked potential to binaural stimulation from the sum of the evoked potential to monaural stimulation. The present study deals with the maturation of the binaural interaction component (BIC) of the middle latency potentials in infants and the difference seen between infants and adults. Eight adults and ten term infants were used in this study. Auditory evoked potentials were obtained for: (1) left monaural stimulation; (2) right monaural stimulation; and (3) binaural stimulation. Individual binaural interaction waveforms were derived as described above. A shift of the major component of the binaural interaction between the adult and infant was observed. The largest BIC occurs in the adult at the time of P30 and N40 with a 41% interaction at P30 and a 21% interaction at N40. The morphology of the infant middle latency potentials differs significantly from the adult form by the narrowness of an N13 component and the lack of a clearly defined P30. N40 does not occur in the infant. The major interaction in the infant occurs at the time of N20, showing an interaction of 45%. This difference probably reflects changes in the middle latency components between adults and infants, with the adult showing a predominant P30–N40 component. [Work supported in part by the National Institutes of Health and the National Foundation March of Dimes.]

9:45

W6. The use of paired clicks to examine forward masking in auditory brainstem responses. Lawrence Shotland and Kurt Hecox (Project Phoenix of Madison, Inc., and Waisman Center on Mental Retardation and Human Development, University of Wisconsin–Madison, Madison, WI 53705)

Forward masking was assessed in ten normal and ten sensorineural hearing-impaired subjects using a paired-click paradigm. Auditory brainstem responses (ABR) were obtained using single and paired clicks. The single or click pairs were repeated at 4.9/s while the separator between clicks for click pairs was 5 ms. First or single clicks were presented at 60 dB nHL for the normal listeners and at levels sufficient to produce a 6.0-ms wave V in the hearing-impaired group. The second click was presented at the same intensity, 10 dB, less intense, and 20 dB less intense than the first click. A digital subtraction technique was utilized to visualize the response to the second click in the pair. Differences in response threshold and in the rate of latency of amplitude growth were observed between the hearing-impaired and normal population. Control conditions suggested that these findings cannot be attributed solely to differences in the level of presentation between the two groups. The contribution of adaptation/forward masking to the results found in the hearing-impaired data will be discussed.

10:00

W7. Auditory sensitivity via BAEP and electrocochleography: Method for improved low-frequency region response. Neil Shepard, Laura Brady, and Eugene Potesta (Divisions of Audiology and Otolaryngology, Henry Ford Hospital, 2799 West Grand Boulevard, Detroit, MI 48202)

Difficulty in realizing a repeatable low-frequency region response near threshold for click (low-frequency) or tone-burst stimuli is partially due to the reduction in synchronous discharge of auditory nerve fibers innervating areas of the cochlea most sensitive to low-frequency input. This problem is exacerbated by the use of random or alternating phase onset to reduce electrical and physiological interference. A method shown to eliminate the time smearing effect of random or alternating phase onset for click stimuli using a fixed phase onset with broadband masking and a subtraction technique was applied to tone-burst stimuli for BAEP and EcochG recordings. Preliminary results show the technique to be as applicable to BAEP recordings and tone-burst stimuli as with EcochG and clicks, provided a sufficient level of broadband noise is presented. The success regarding low-frequency response in both normal-hearing and hearing-impaired subjects will be presented.

10:15

W8. Clinical validation of special hearing-impaired norms for BAEP interpretation. Neil T. Shepard, Mary-Jo Burtka, and Robert G. Turner (Division of Audiology and Otology Research, Henry Ford Hospital, 2799 West Grand Boulevard, Detroit, MI 48202)

A model predicting the effects of high-frequency hearing loss of cochlear origin on the BAEP [N. T. Shepard and J. C. Webster, J. Acoust. Soc. Am. Suppl. 1 74, S40 (1983)] was used to develop criteria based on degree and configuration of loss of sensitivity for interpreting clinical BAEP testing. Two and one-half years of experience with these criteria were reviewed by a retrospective study of 389 patients tested during that period. Validation of the model was indicated by an 8% increase in specificity of the test with no change in sensitivity when use of the special criteria were compared to criteria based solely on normal-hearing subjects in the same clinical population. Also discussed will be overall BAEP performance, an explanation of high false positive rate based on population mix, and improved criteria for test enhancement.

10:30

W9. Auditory middle latency and steady-state responses in infants. David R. Stapells (Albert Einstein College of Medicine, Kenn 825, 1300 Morris Park Avenue, Bronx, NY 10461) Jamie A. Costello, Donna Smith, Scott Makeig, and Robert Galambos (Children's Hospital Research Center, 8001 Frost Street, San Diego, CA 92123)

The effects of stimulus rate, intensity, and tonal frequency on the auditory middle latency and steady-state responses (MLR/SSR) were investigated in 29 normal infants (1–36 months) and in eight normal adults. Adultlike MLRs (10/s) were recordable in only 5 infants, 14 demonstrated responses with "immature" morphology, and ten demonstrated no MLR components after Na. Compared to adults, infant SSRs were lower in amplitude and demonstrate different scalp distributions. Unlike the adult data, no consistent amplitude peak was seen across rate, and phase coherence increased up to highest rate tested (59/s). Infants required higher stimulus intensities to produce adultlike SSR phase coherence values: 83% of the recordings to 70-dB nHL tones presented at 43.4/s showed significant values ($p<0.01$), decreasing to 35% at 40 dB nHL. These results are very different from those we previously reported for adults [Stapells, Makeig, and Galambos, J. Acoust. Soc. Am. Suppl. 1 77, S66 (1985)], and suggest that infant MLRs and SSRs undergo complex changes with maturation. [Work supported by DRF and NIH.]

10:45

W10. Intensity discrimination: Relation of auditory-nerve activity to psychophysical performance. Bertrand Delgutte (Eaton-Peabody Laboratory of Auditory Physiology, Massachusetts Eye and Ear Infirmary, Boston, MA 02114 and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

In order to test the notion that saturation of auditory-nerve fibers limits psychophysical performance in intensity discrimination at high stimulus levels, intensity difference limens (DL) of single auditory-nerve fibers in anesthetized cats were measured for tones at the CF, using stimulus paradigms and detectability measures similar to those of psychophysics. The physiological DL reaches a minimum in the range of stimulus levels where discharge rate increases rapidly. In a narrow range of levels around the minimum, single-fiber DLs approach psychophysical DLs. An optimum-processor model that combines intensity information from an array of 30 000 auditory-nerve fibers with a realistic threshold distribution predicts intensity DLs that are well below psychophysical DLs over a 90-dB range of levels. This result implies that psychophysical performance is not limited by saturation in auditory-nerve fibers, but by central factors. Suboptimal processors that give more weight to intensity information from high-threshold fibers than to information from low-threshold fibers can closely predict psychophysical DLs for both tones and broadband noise over a wide range of stimulus levels. Moreover, such processing schemes, which are based on average discharge rates, provide a stable representation of the spectra of speech sounds with respect to variations in intensity. [Work supported by NIH.]

11:00

W11. Cochlear tuning characteristics in infants and adults. J. Y. Bargones and E. M. Burns (Department of Speech and Hearing Sciences, JG-15, University of Washington, Seattle, WA 98105)

Cochlear tuning characteristics were estimated by obtaining suppression tuning curves (STCs) for spontaneous otoacoustic emissions (SOAEs) in infant and adult subjects. STCs were derived using 8 to 12 suppressor frequencies per SOAE. Infants were tested at approximately 3 weeks, 2 months, and 3 months of age. Adults were tested three times over a period of 3 months. Quantitative analyses include Q-10, Q-20, slopes of the high- and low-frequency segments of the tuning curve and tip frequency. Results indicate that cochlear tuning properties in infants approximate those found in adults; however, greater variability is observed for repeated measures of infants STCs. The results are discussed with respect to the development of frequency selectivity. [Work supported by NINCDS.]

11:15

W12. Frequency selectivity of the active cochlear filters inferred from otoacoustic emissions evoked by steady-state tones. Pierre L. Divenyi (Speech and Hearing Research, V.A. Medical Center, Martinez, CA 94553 and Department of Speech and Hearing, University of California, Santa Barbara, CA 93106)

Spontaneous otoacoustic emissions have been shown to represent an oscillatory process rather than amplification of a narrow-band noise centered at the emission frequency: Instantaneous amplitudes of the emission have the bimodal distribution characteristic to oscillations (W. Bialek and H. P. Wit, Phys. Lett. 104A, 173–178). The present paper attempts to demonstrate that similar bimodal distributions may be observed when the response to a low-level (<30 dB SPL) steady-state tone is recorded near the tympanic membrane and the energy at the tone frequency is eliminated from the recording. At either side of the stimulus frequency, points at which the narrow-band instantaneous amplitudes shift from a unimodal to a bimodal distribution may be mapped into bandwidths that are comparable to those deriving from physiological measurements of the tip of the tuning curve. [Work supported by the Veterans Administration.]

11:30

W13. Otoacoustic emissions and cochlear pathology. J. L. Grizzle, P. L. Divenyi, and H. J. Simon (Speech and Hearing Research, Veterans Administration Medical Center, Martinez, CA 94553)

Otoacoustic emissions, evoked by clicks and brief tone bursts, were measured in normal-hearing subjects and patients with sensorineural hearing loss. The emissions were recorded using an electret microphone assembly inserted in the meatus. The stimulus was delivered through a transducer attached to a 1.3-mm plastic tube terminating at approximately 1 cm from the eardrum. Spectral analysis of the click-emitted emissions showed discontinuities similar to those observed in spontaneous emission recordings. In addition, the comparison of emission and audiometric data in the impaired ears will be reported. [Supported by the Veterans Administration.]

WEDNESDAY MORNING, 10 DECEMBER 1986 SALON E, 8:30 A.M. TO 12:30 P.M.

Session X. Speech Communication III: Intonation, Stress, Synthesis by Rule

John J. Godfrey, Chairman
Texas Instruments Company, MS 238, P.O. Box 225474, Dallas, Texas 75265

Contributed Papers

8:30

X1. Some phonetic characteristics of speech produced in noise. Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701), Thomas J. Moore (Armstrong Aerospace Medical Research Laboratory, Wright-Patterson AFB, OH 45433), and Beverley Gable (Department of Psychology, Ohio University, Athens, OH 45701)

This study is concerned with changes in the acoustic–phonetic structure of isolated words produced when the talker is in a high-noise environment and/or is wearing equipment that produces an obstruction to articulation, e.g., an oxygen mask. The speech of four young males was recorded in three conditions: quiet (laboratory recording conditions), while listening to white noise at 95 dB SPL over earphones, and while listening to the same noise levels introduced into the earphones of a standard Air Force helmet equipped with an oxygen mask. Speech produced while listening to noise over earphones alone showed the expected increases in fundamental frequency and amplitude. In addition, the duration and formant structure of vowels, and to a lesser degree the timing of consonantal gestures, changed when speech was produced in noise. Preliminary indications are that speech produced while wearing a helmet and oxygen mask was minimally affected by noise introduced over the headphones. [Supported in part by AFOSR.]

8:42

X2. Acoustic characteristics of emphasis. John J. Godfrey and Joshua M. Brodsky (Texas Instruments, M.S. 238, P.O. Box 225474, Dallas, TX 75265)

The fundamental frequency, intensity, and duration of the stressed syllable nucleus are three of the principal cues used to convey the amount of emphasis intended on a particular word in English. In this study we examined how these three characteristics covaried in a set of 800 utterances produced by eight speakers. The sentences, which were designed and collected for another experiment [Steele, 1986], each contain one disyllabic name spoken with contrastive stress and another which is postnuclear and deaccented. The speakers were asked, in effect, to vary the degree of emphasis on the name with contrastive stress over a wide range. Measurements of the $f0$, intensity, and duration of the stressed syllables in both positions suggest that individual speakers chose different "strategies" for employing these cues in combination. Pitch elevation was used most consistently over the entire range.

8:54

X3. Syllabic constraints on P centers. Mary R. Smith (A.P.U., 15 Chaucer Road, Cambridge CB2 2EF, England)

The location of P centers in monosyllables is affected by changes in durations of phonetic segments. Four natural ways to affect duration have been studied previously: phonetic changes in syllable onset, nucleus, and coda, and changes in speaking tempo. Syllable durations are also affected by the presence of other syllables within a word and the resultant stress patterns. In this study the effects of syllable compression and lengthening on P-center location are examined, together with effects of stress pattern, in natural productions of morphologically derived and inflected forms of base words such as *prove* and *stead*. The P centers are describable with reference to the stressed vowel in polysyllables. The previously observed asymmetry of effect between syllable onsets and rimes appears not to extend to prefixes and suffixes beyond the effects such affixes have on the stressed syllable segment durations.

9:06

X4. Vowel quality and word stress in native and non-native English. Joann Fokes (School of Hearing and Speech Sciences, Ohio University, Athens, OH 45701) and Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701)

In a recent study [Fokes and Bond, J. Acoust. Soc. Am. Suppl. 1. **79**, S27 (1986)], we reported that non-native speakers of English lengthened unstressed syllables and shortened stressed syllables of three and four syllable words, such as *confession* and *combination*, in comparison to native English speakers' patterns. While all speakers were not consistent in their use of either fundamental frequency or amplitude in differentiating stressed from unstressed syllables, the non-native speakers were much more variable. In an attempt to clarify differences in speech patterns between these two groups of speakers, we measured the first two formants of the first and second syllable of the same set of words as produced in isolation and in sentence context. The $F1$ and $F2$ values of the native English speakers were as expected and differed little from isolation to sentence context. The non-native English speakers varied $F1$ and $F2$ from isolation to sentence context but with no discernible pattern. The formant measurements will be discussed both in terms of intended and perceived vowels.

9:18

X5. Co-occurrence of allophonic forms. Jared Bernstein, Gay Baldwin, and Hy Murveit (Speech Research Program, S.R.I. International, Menlo Park, CA 94025)

Recent analyses of the allophonic variants of phonemes in particular environments often seem to assume one or more of the following: (1) The proportions of variants encountered in a multispeaker sample represent an "irreducible" statistical component of phonology; (2) these proportions predict the likelihood of encountering these same allophones in new material; (3) the probability of encountering a particular allophone of some phoneme is independent of the observed allophones of other phonemes nearby. These related assumptions are questioned on theoretical and practical grounds, using transcribed data from 630 speakers reading two sample sentences. The frequencies of occurrence of all the allophones of certain phoneme tokens in the sample sentences were measured across all speakers; then the conditional co-occurrence of all the allophones of certain phoneme pairs in the sample sentences were analyzed. For example, first the occurrence of a flap in the words "suit in" was counted across all speakers; then the occurrence of a flap in "suit in" was counted for only those speakers who deleted /t/ in "don't ask," and so on. Comparing these two kinds of analysis has implications for theories of variation. The application co-occurance analysis in speech recognition will be illustrated. [Work supported by DARPA.]

9:30

X6. Creak as a sociophonetic marker. Caroline G. Henton (Linguistics Program, University of California at Davis, Davis, CA 95616)

Previous investigations into creak (vocal fry or pulse phonation) concentrated on two aspects: establishment of the physiological, acoustic, or perceptual nature of creak compared with other phonations; or separation of its characteristics in normal speech from pathological voices. The approach here is entirely different. Taking the phonetic and linguistic incidence of creak in normal speech in two accents of English as a starting point, quantitative description of its occurrence is provided. Data are taken from large numbers of speakers producing meaningful utterances in a recording task unrelated to voice quality. Support is given to the previous (unquantified) notion that creak accompanies utterance-final low falls and may be a turn-relinquishing signal. In addition, important cross-accent and cross-sex differences in the usage of creak are revealed. The results indicate that future descriptions of male–female voice quality differences should acknowledge creak as an important sociophonetic marker. Furthermore, phonological accounts of accents should incorporate information about a community voice quality. Last, clinicians will benefit from a quantitative account of creak in "normal voice" and may be led to adjust their parameters of abnormality accordingly.

9:42

X7. Discrete acoustic analysis of intonation patterns in controlled contextual settings. Marc E. Pratarelli (Speech Science Research Laboratory, 3375 South Hoover, Los Angeles, CA 90007)

Acoustical and temporal correlates of intonation patterns in eight semantically distinct sentences made up of the same three words, "*Bob bit Todd*," were quantified, verified perceptually, and analyzed statistically for degrees of variability. The phrases consisted of four declaratives and four interrogatives. Three of each group of four systematically varied the stress component on each of the three words. The fourth phrase type was a neutral version with no particular emphasis on any member. A single speaker produced ten replications over three successive days in response to experimenter queries; listener judgments were used to verify the contextual appropriateness in all 80 utterances. Duration, fundamental frequency, and intensity were measured systematically to study both the variation of each and the interrelations between them. Several graphs are presented that depict the movement of the means and standard deviations of the acoustic and temporal correlates, and pitch contours are presented which depict the degree of stability of instability of the correlates. Canonical correlations and regression analysis predictably isolated the magnitude of fundamental frequency stability in the interrogatives but not the concomitant high degree of variability in the same utterances. An index of what "*normal variation*" may be, and its impact on recognition, is discussed. [Work supported by the House Ear Institute.]

9:54

X8. Intrinsic pitch of vowels in read speech at different speech rates. Kazue Hata (Department of Linguistics, University of California at Berkeley, Berkeley, CA 94720)

Although there is abundant evidence for systematic differences in the average $F0$ of individual vowels (so-called "intrinsic pitch"), other things being equal, it has been claimed that these differences disappear in connected speech [N. Umeda, J. Acoust. Soc. Am. 70, 350–355 (1981)]. In the hope that it could be possible to identify the other things which, when not equal, influence intrinsic vowel pitch, how it is affected by speaking rate and emphasis (contrastive stress) was examined. Ten speakers of American English (five male and five female) read sentences containing "Leo" and "Lolly" in the target sentence-medial position with and without emphasis at three different rates. A preliminary analysis of the data reveals that, although these factors influence the intrinsic pitch differences, namely, emphasis amplified it and fast rate attenuated it, the effect was preserved on the average under all conditions.

10:06–10:30
Break

10:30

X9. Effect on F_0 of the linguistic use of tense and lax phonation. Ian Maddieson and Susan Hess (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

Laver [*Phonetic Description of Voice Quality* (Cambridge U.P., London, 1980)] remarks that "there is a strong possibility that in tense voice the pitch range will be higher than in lax voice," but notes that adjustments can be made to compensate for this tendency. Three languages with phonological contrasts between syllables with tense (somewhat creaky) and lax (somewhat breathy) phonation were examined to see how greatly F_0 differed between such syllables. It was expected that compensatory adjustments would be made in tone languages in order to preserve contrasts between tones, but not in nontonal ones. In Wa and in Northern Yi, after most types of consonants were examined, there was no significant F_0 difference between tense and lax syllables. In Jingpho, F_0 at vowel onset was reliably higher in tense syllables, but there was no difference at vowel offset. Since Wa is nontonal, while Yi and Jingpho have similar tonal systems, our expectations were not met. However, a potential role for a lax/tense phonation contrast in tonogenesis and tone splitting was revealed. [Work supported by NIH.]

10:42

X10. Fundamental frequency variability in the oral reading of untrained English speakers. James R. Solomon (Department of Communicative Sciences and Disorders, California State University, Hayward, CA 94542) and Eric J. Soares (Department of Marketing, California State University, Hayward, CA 94542)

A characteristic common to the speech of virtually all individuals with dysarthria (a motor speech disorder) is "monopitch" or "monotone"; yet many such patients possess variations in fundamental frequency ($F0$) that are perceptually apparent and instrumentally demonstrable. In an attempt to establish reference data for the degree of $F0$ variability in the oral reading of normal English speakers, 76 randomly selected, naive native speakers of American English were asked to read aloud a standard paragraph used to assess contextual speech in both normal and deviant speakers (the "Grandfather" passage) as part of a study correlating perceptual and acoustic measures of expressiveness in speaking. Selected sentences of the tape-recorded paragraphs were visually inspected on a real-time $F0$ analyzer (the Visi-pitch), with measurements made (wherever possible) at regularly occurring intervals in the sentence. Mean $F0$'s for each sentence were then calculated as well as standard deviations, which reveal the degree of departure from a theoretical monotone (s.d. $=0$). These data along with details of instructions to subjects and acoustical analysis techniques used will be presented.

10:54

X11. Nuclear accent $F0$ peak location: Effects of rate, vowel, and number of following syllables. Shirley A. Steele (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Reported here are results from an investigation of the alignment of speech segments with the fundamental frequency ($F0$) contour at the end of an intonational phrase. Findings suggest that the $F0$ peak on the main sentence stress (nuclear stress) occurs at some proportion of the length of the vowel, instead of at a fixed distance into the vowel. The vowel may be shortened by a faster speaking rate, an intrinsically shorter vowel, and added post-nuclear syllables. The $F0$ peak delay, however, is shortened by faster speaking rate and intrinsically shorter vowels, but lengthened or unaffected by added syllables. Thus vowel length and post-nuclear syllables interact to determine the location of the $F0$ peak.

11:06

X12. On the interaction of tone and intonation in Mandarin Chinese. Chilin Shih (Department of Linguistics and Artificial Intelligence Research, AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

Intonation in many languages, including English, is realized most effectively by changes in fundamental frequency. It is important to investigate whether this relation holds true in a tone language, where $F0$ variation may affect lexical meaning. This paper reports the interaction of tones and intonation in a tone language, Mandarin Chinese, and calculates the effects of various intonation patterns as well as focus structures. The result is implemented and tested in a Chinese text-to-speech system. In general, the interrogative pattern in Mandarin Chinese is characterized by a high reference line and a high boundary tone at the end of a sentence. However, the resulting tone shape is not always rising as a result of the lexical tones. Focus is realized mainly by increasing the pitch range, but the exact implementation is complicated by some local tonal effects such as target shift. This paper will also explain the interaction of catathesis (pitch lowering triggered by specific tonal context) and focus in the direction of prosodic structure.

11:18

X13. Measuring the contribution of formant frequency targets, bandwidths, and transitional timing to the perception of a vowel idiolect. James T. Wright, Bathsheba J. Malsheen, and Melanie Yue (Speech Plus, Inc., 461 North Bernardo Avenue, Mountain View, CA 94043)

One goal in speech synthesis-by-rule is to replicate the idiolectal properties of human speakers in order to produce more natural and distinctive synthetic voices. The spectral properties of vowel idiolects can be expected to vary in formant frequencies, timing of transitions, bandwidths, and source spectra. These characteristics can be combined into a model that specifies vowels in terms of formant targets for the nuclei and offglides of the vowels, the percent of the duration at the midpoint of transitions, and the duration of transition [J. Allen, *From Text to Speech: The MITalk System* (1986)]. These parameters were measured in stressed monosyllables for diphthongs and [Vr] sequences in a frame sentence. Data from an AX discrimination test will be presented which indicate the relative contribution of each model parameter to simulating the natural vowels.

11:30

X14. Prosodic variants of syllable templates for speech synthesis. Marian Macchi, Jennifer O'Brien, and Lynn Streeter (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

In many speech synthesis sytems based on concatenation of prerecorded units, the same segmental template is used in different prosodic environments, typically with simple manipulations of acoustic parameters. The present experiment investigated how successful simple techniques are for syllable-based synthesis. English speakers produced CVC and CV syllables with /p,b,w/ and /i,a,ay/, in isolation and in polysyllabic reiterant phrases (e.g., /bayb bayb bayb bayb/) with different stress and intonation patterns. Pitch tracking the LPC analysis were performed, and syllable templates were excised. Hybrid phrases were created by replacing a syllable template in one prosodic environment with a template for the same syllable from another prosodic environment, substituting the appropriate duration, fundamental frequency, and amplitude. Listening tests showed that natural-sounding hybrid unstressed syllables can be constructed from templates from stressed original syllables, but not vice versa. Also, hybrids should retain the syllable position of the original templates (e.g., phrase-initial, phrase-final). These results are useful for determining which manipulations produce natural-sounding speech, and deciding whether multiple templates for the same segmental unit are necessary.

11:42

X15. Effects of cue impoverishment on intelligibility and naturalness of synthesized velar stops. Bathsheba J. Malsheen, James T. Wright, and Melanie Yue (Speech Plus, Inc., 461 North Bernardo Avenue, Mountain View, CA 94043)

In a previous test of two MITalk-based synthesizers [B. J. Malsheen, J. T. Wright, M. Yue, and M. Peet, J. Acoust. Soc. Am. Suppl. 1 **79**, S25 (1986)], we found that the intelligibility of velar stops degraded more for one system than the other in a simulated telephone bandwidth condition. We hypothesized that this degradation was due to missing secondary cues normally present in human velar productions. In order to test this hypothesis we examined short-time spectra of the synthesized velar bursts and compared them with those produced by a male human speaker. We found that the synthesized velars lacked a number of acoustic cues present in the human productions. Secondary high-frequency energy peaks for initial and final velars before and after nonfront vowels were missing, primary peaks for front vowels were lower in frequency than those apparent in the human spectra, and primary peaks for nonfront vowels were higher. Velar stops were then resynthesized on the basis of the human model spectra. Preliminary results show that the more fully cued velars sound less "chirpy" and more natural both in normal and telephone conditions, and that intelligibility has improved.

11:54

X16. Synthesis of falling nuclear pitch accents. Mark Y. Liberman and Shirley A. Steele (AT&T Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

A falling ("declarative") nuclear pitch accent has a simple shape: an upward pitch obtrusion on the accented syllable, a fairly abrupt fall to near the bottom of the speaker's range, and a more gradual fall to the end. The synthesis algorithm in Anderson, Pierrehumbert, and Liberman (1983) decomposes this pattern into a H^* pitch accent followed by an L phrase accent and an L boundary tone. It devotes four parameters to the realization of the H^* accent (amount of rise, target $F0$ height, target duration, target time) and two parameters to the L phrase accent (target $F0$, target time). The parameters depend in part on properties of the accented syllable (e.g., prominence) and in part on characteristics of larger phrases (e.g., pitch range). The resulting time/$F0$ pairs are subject to interpolation and smoothing. This paper improves on the naturalness of the algorithm by incorporating findings from a recent study by Steele. In particular, the principles that determine the timing, magnitude, and alignment of the H^* $F0$ rise and of the H^* to L $F0$ fall have been improved.

12:06

X17. Stressing English noun compounds correctly. Richard W. Sproat and Mark Y. Liberman (Bell Laboratories, 2D-518, 600 Mountain Avenue, Murray Hill, NJ 07974)

We describe a program for assigning correct stress contours to noun compounds in English. It makes use of two kinds of knowledge: idiosyncratic knowledge about the stress behavior of various compound types and general knowledge about English stress rules. As an example of the first, place names such as *Madison AVENUE* generally have rightward stress, but the word *street* induces left-hand stress: *MADISON street*. Semantic relations between the component nouns are also important: We know that, if the left-hand member is a grammatical object of the right-hand member, stress is leftward (*DOG catcher*). Our program makes use of such idiosyncratic knowledge in determining stress by deducing the type of compound and the likely relations between the members. As an example of the second kind of knowledge, our program knows the "rhythm rule" whose consequence is that while stress in *Tom PAINE* is on the right, stress in *TOM Paine AVENUE* is stronger on *Tom* than on *Paine*. The obvious importance of such a program to speech synthesis is discussed.

12:18

X18. Phonetic dependencies in syllables with full and reduced vowels. Candace Kamm and Marian Macchi (Bell Communications Research, 435 South Street, Morristown, NJ 07960)

Speech synthesis using demisyllable concatenation relies on the assumption that initial and final demisyllables are phonetically independent of each other, aside from the need for smoothing the boundary. The success of demisyllable concatenation reported by Lovins, Macchi, and Fujimura [J. Acoust. Soc. Am. Suppl. 1 **65**, S130 (1979)] suggests that any phonetic dependencies that exist are not perceptually salient, at least for phrase-final syllables containing stressed vowels. To asess the independence assumption for naturally spoken English CVC syllables, the influences the initial consonants on syllable-final events and of final consonants on syllable-initial events were examined for long-duration syllables containing full vowels and shorter-duration syllables containing full and reduced vowels. Reduced syllables showed larger variation in formant frequencies at vowel offset across initial consonants than full-vowel syllables, including those with comparable vowel duration. Similarly, reduced syllables showed larger variation in formant frequencies at vowel onset across final consonants than syllables containing long-duration full vowels. To maintain the contextual variation in reduced vowels in demisyllable synthesis, it may be necessary to devise more complex smoothing rules or to include several versions of reduced-vowel demisyllables in the inventory.

WEDNESDAY MORNING, 10 DECEMBER 1986 SALONS G AND H, 9:00 A.M. TO 12:00 NOON

Session Y. Underwater Acoustics IV: Range-Dependent and Under-Ice Propagation

Michael J. Buckingham, Chairman
Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Contributed Papers

9:00

Y1. Characterization of range-dependent shallow water waveguides. George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543), Ferdinand J. Diemer, and Peter H. Dahl (MIT/WHOI Joint Program in Oceanography/Oceanographic Engineering, Woods Hole, MA 02543)

A technique for characterizing range-dependent shallow water waveguides is described. The method consists of determining the beamformed output of a horizontal array over short apertures for signals due to a cw point source. By modeling the acoustic field locally as a sum of damped normal modes and using Prony's method to perform the beamforming, the local modal structure of the waveguide can be resolved. As a result, the modal composition of the waveguide as a function of range can be determined and interpreted in terms of range-dependent mode theories (e.g., adiabatic mode theory). In addition to identifying important propagation characteristics such as mode cutoff, the method can be used to determine range-dependent acoustic properties of the bottom. Examples of the application of the technique to the case of propagation in a wedge-shaped ocean are presented. [Work supported by ONR.]

9:15

Y2. High-frequency acoustic propagation in a range-dependent guiding-to-antiguiding ocean channel transition. L. B. Felsen (Polytechnic University, Route 110, Farmingdale, NY 11735) and T. Ishihara (The National Defense Academy, Hashirimizu, Yokosuka, 239, Japan)

Variations in the ambient physical parameters in the ocean may be such as to change a depth-dependent profile with weak range dependence from guiding to antiguiding. This paper examines conversion of an initially well-guided high-frequency adiabatic mode into a nonguided sound field after passing through the transitional profile region. A theory is developed which smoothly patches a parabolic equation for the transitional domain onto trapped adiabatic mode fields on the guiding side and onto leaky modes on the nonguiding side. The parabolic equation is implemented numerically where it cannot be reduced to the simpler asymptotic forms. Numerical results for the dominant mode in a model profile confirm the analytical features of the theory, and reveal clearly the phenomenology of initial confinement in a surface duct, with subsequent detachment and "beaming" into deep water after passing through the transition to the antiguiding environment. [Work supported by ONR.]

9:30

Y3. VLF pulse propagation in range-dependent geoacoustic waveguides. R. Stephen and M. Holzrichter (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

The finite difference method is used to study VLF pulse propagation (peak frequency of 10 Hz) in shallow water waveguides (100-m water depth) with range-dependent depth and geoacoustic parameters out to ranges of 5 km. The method solves the full two-way elastic wave equation in space and time using an explicit scheme based on centred finite differences. Compressional and shear velocity and density can be varied arbitrarily in a two-dimensional grid. The models represent continental margin environments with both upslope and downslope propagation. Propagation can be studied in range-time space, in snapshots of the wave field at given instants in time and in frequency wavenumber space. In soft bottom environments where the shear wave velocity of the bottom is less than the compressional wave velocity in the water, energy is continually leaking into converted shear waves in the bottom and there are no "perfectly trapped modes." However, for sources and receivers near the seafloor, Stonely waves are observed with velocities near the shear wave velocity. For sources over the continental shelf and receivers over the continental slope, the numerical experiments predict two new wave phenomena. Compressional ground wave arrivals are enhanced by focusing at the shelf break and water wave modes do not adapt immediately to changes in water depth. In the latter case, the water wave modes set up in the shallow water waveguide persists almost unaffected out to 2 km beyond the shelf break. Reflections of energy propagating back to the source from the shelf break are also observed.

9:45

Y4. Parabolic equation modeling of normal mode propagation in a wedge. Finn B. Jensen (SACLANT ASW Research Centre, 19026 La Spezia, Italy) and C. T. Tindle (Physics Department, University of Auckland, Auckland, New Zealand)

The parabolic equation model has been used to investigate the behavior of the "wedge modes" described in the experimental results of Hobaek, Tindle, and Muir [J. Acoust. Soc. Am. Suppl. 1 78, S70 (1985)] for propagation in a shallow water wedge. The wavefronts of wedge modes are not vertical, but are curved into arcs of circles centered on the wedge apex. Individual wedge modes can be examined by simulating a curved line source. A wedge mode propagates without coupling to other modes. By contrast, if the line source curvature is omitted it is not possible to excite a pure mode, and strong mode interference is observed. As wedge modes propagate upslope through their cutoff depth, their energy is dumped into the bottom as a beam. Any coupling to other modes as a mode passes through cutoff is very small and may be ignored.

10:00

Y5. Sound propagation in three dimensions using a new numerical algorithm. D. Lee, P. D. Scully-Power, G. Botseas (Naval Underwater Systems Center, New London, CT 06320), and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12180-3590)

Recently, a numerical algorithm was developed for solution of a three-dimensional, wide-angle parabolic approximation to the Helmholtz equation [D. Lee, Y. Saad, and M. Schultz, Proc. First IMACS Symp. Comp. Acoust. (to appear)]. The method is both stable and efficient, since each range step requires solving only two tridiagonal systems. Numerical computations have been performed for test examples and model problems, and the accuracy and efficiency of the method have been illustrated. Extensions and improvements in the algorithm are reported here. These include implementations on parallel processors and on supercomputers, and modifications for switching between two- and three-dimensional calculations. The latter feature permits bypassing the full three-dimensional capabilities of the code where environmental conditions are sufficiently simple. Data requirements for three-dimensional computations using this new algorithm are discussed, in connection with longer range propagation through models of mesoscale variations such as fronts and eddies. [Work supported by NUSC and ONR.]

10:15

Y6. Time series simulation in a sloping bottom environment using ray theory. Evan K. Westwood (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8129), H. Hobaek (Department of Physics, University of Bergen, Bergen, Norway), and C. T. Tindle (Physics Department, University of Auckland, Auckland, New Zealand)

A method for simulating time series in shallow water using ray theory has been extended to an isovelocity wedge with a penetrable bottom. Beam displacement at the water–bottom interface is included, and the resulting caustics are treated with correction factors. Simulated time series compare favorably with time series measured in an experimental, scale model tank using an 80-kHz pulse as a source signal. Mode extraction from sets of time series at different water depths works well on both simulated and experimental data. Phenomena such as wave front curvature, upslope mode cutoff, and downslope mode capture are investigated using the ray model. [Work supported by Independent Research and Development, Applied Research Laboratories, The University of Texas at Austin.]

10:30

Y7. Attenuation of the modes of propagation in an homogeneous floating ice plate. Peter J. Stein[a] (Department of Ocean Engineering, MIT, Cambridge, MA 02139)

The characteristic equation for plane-wave propagation in a homogeneous floating ice plate was solved numerically to determine the phase speeds and attenuations of the first- and second-order modes. While discussion of the phase speeds of the first-order modes constitutes a review, the discussion of the second-order modes, along with the attenuation characteristics of the modes when ice absorption is introduced, gives new insight into which modes might be observed in Arctic pack ice. Only the flexural and longitudinal waves, which exist below a frequency-ice thickness product of 300 Hz-m, propagate with losses less than 0.1 dB/m in ice less than 3 m thick. This is important to the study of noise from nearby ice events. Results of using a nearby explosive charge to measure the ice longitudinal wave speed and attenuation are given. The ice loss in the 40-Hz region was found to be approximately a factor of two higher than expected from current empirical absorption values. This may be important to understanding losses from ice interaction in long range propagation. [Work supported by ONR.] [a] Present address: Atlantic Applied Research Corp., 129 Middlesex Turnpike, Burlington, MA 01803.

WEDNESDAY AM

10:45

Y8. On the controversial leaky Rayleigh wave at a water/ice interface. Jacques R. Chamuel (Sonoquest/Advanced Ultrasonics Research, P. O. Box 153, Wellesley Hills, MA 02181)

To the author's knowledge, no experimental data have been published to date isolating and demonstrating the existence of the controversial leaky Rayleigh wave originating at a water/ice interface. In this paper, ultrasonic laboratory results are presented demonstrating the existence of a leaky Rayleigh wave at a water/ice interface not meeting Brower's (1979) existence condition. The measured phase velocity of transient leaky Rayleigh waves is about 8% higher than the free Rayleigh wave velocity from the air/ice interface. The refracted shear wave amplitude is very small compared to the leaky Rayleigh wave. The received signal is dominated by the leaky Rayleigh wave when the receiver is simultaneously close to the source and at a distance of about one Scholte wavelength from the interface. Substantial energy is associated with the leaky Rayleigh wave indicating that it may be important to account for the contribution of the leaky Rayleigh wave in the problem of scattering from rough water/ice interfaces. The leaky Rayleigh wave is very susceptible to the presence of surface cracks in the ice. The reported findings provide physical insights into arctic acoustics and may contribute to the interpretation of arctic acoustic data. [Work supported by ONR.]

11:00

Y9. The influence of thermohaline steps on under-ice acoustic propagation: A simulation study. Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529-5004)

A simulation study of the influence of thermohaline steps on under-ice acoustic propagation has been made using a surface representation of hard layered ice with a water-to-ice transition region and a "stair-step" sound velocity profile (SVP) in the water. The ice surface was modeled using parallel layers of varying thicknesses, each layer being homogeneous in density, compressional velocity and attenuation, and shear velocity and attenuation. A water-to-ice transition region [S. A. Chin-Bing, J. Acoust. Soc. Am. Suppl. 1 **78**, S57 (1985)] was included to allow a gradual transition from the water to the multilayered ice. The water region contained the stair-step SVP that is often found in regions containing thermohaline steps [S. A. Chin-Bing and D. B. King, J. Acoust. Soc. Am. Suppl. 1 **76**, S84 (1984)]. Results indicate that the transmission loss structure is affected by both the stair-step SVP and the water-to-ice transition region.

11:15

Y10. Seismic propagation velocity measurements in arctic sea ice. J. M. Ozard and G. H. Brooke (Defence Research Establishment Pacific, FMO, Victoria, British Columbia, V0S 1B0, Canada)

It has been predicted that the seismic propagation velocities of sea ice affect the propagation of sound in ice covered arctic waters. Since very little data are available on propagation velocities, a series of *in situ* measurements of propagation velocities and associated density, temperature, and salinity profiles have been made in sea ice. Seismic energy from predominantly shear or compressional wave sources was propagated over ranges of a few hundred meters to two three-component geophone arrays. Propagation paths in smooth and in slightly rough annual ice were selected. Plate, flexural, and shear wave arrivals were clearly identified from their polarizations and particle velocities. A reduction in plate and flexural wave velocities was observed in the rough annual ice compared to the smoth annual ice.

11:30

Y11. A range-dependent normal mode model with full mode coupling. E. Richard Robinson (Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320) and David H. Wood (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. We use full mode matching at the interfaces to compute the acoustic field. Our approach is similar to that used by Evans and Gilbert ["Acoustic propagation in a refracting ocean waveguide with an irregular interface," Comp. Math. Appls. **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. We implement within the generic sonar model because this allows for considerable flexibility in updating developments and the choice of supporting submodels.

11:45

Y12. Limitations of sound propagation in the ocean: The curtain effect. D. G. Browing, J. J. Hanrahan, R. J. Christian (Naval Underwater Systems Center, New London, CT 06320), and R. H. Mellen (PSI-Marine Sciences, New London, CT 06320)

Although initially very high, the rate of spreading loss decreases rapidly with range, while the rate of attenuation remains constant for a given frequency. At increasing ranges the two loss curves cross, with attenuation becoming the dominate mechanism. This results in a "curtain effect" due to rapidly increasing propagation loss. Examples are given of convergence zones obtainable as a function of frequency for various oceans and of the transition between near range and distant ambient noise. [Work supported by NUSC.]

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

J. C. Barton, Chairman S2
Caterpillar Tractor Company, Research Department, 100 N. E. Adams, Peoria, Illinois 61629

G. Booth, Chairman, Technical Advisory Group for ISO/TC 108
200 Clark Avenue, Brandford, 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) including plans for the forthcoming meeting of ISO/TC 108, scheduled to take place in Washington, DC from 30 March to 10 April 1987.

Session Z. Architectural Acoustics IV: Theory, Measurements, and Materials

Alfred C. C. Warnock, Chairman
Division of Building Research, National Research Council of Canada, Montreal Road, Ottawa, Ontario K1A 0R6, Canada

Contributed Papers

1:45

Z1. Vortex modes in rooms. Richard V. Waterhouse (Code 1940.2, David Taylor Naval Ship R&D Center, Bethesda, MD 20084-5000)

The existence of a circulation of sound energy in an enclosure of square cross section was demonstrated on theoretical grounds by Preston Smith in 1963. Recent work has confirmed experimentally the flow of sound energy in a vortex pattern in various sound fields. Here it is shown that for any rectangular enclosure (e.g., a room or duct) having reflecting walls with two dimensions in the ratio of any two integers, there exists an infinite set of vortex modes, whose frequencies form a harmonic series. Also, for an enclosure in the form of a rigid circular cylinder, there exists infinite sets of vortex modes whose frequencies depend on the extreae of the Bessel functions J_m. Energy streamlines are given for some of the above modes, and their practical significance is discussed.

2:00

Z2. Impulse-response and transfer-function measurements in rooms by *m* sequence cross correlation. E. Paul Palmer, Rodney D. Price, and Steven J. Burton (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

The impulse response and transfer function between a sound source and receiver in a room can be measured using cross-correlation techniques with a maximum-length sequence as the signal source. A graphic representation of time evolution of direct sound, echoes, scattering, etc. (via the impulse response) and of spectral effects (via the transfer function) provide tools for acoustical design. The methods are illustrated for diffraction around an edge in an anechoic chamber. "Textbook"-like interference patterns are not seen because of the broadband nature of the signal. Instead, a quantitative measure of scattering, appropriate for speech and music, is obtained. Quantitative spectral measures are obtained using the transfer function. In an ordinary room, the effects of changing reflecting, scattering, and absorbing walls and objects and of displacing source and receiver are graphically displayed. The effects of modifying and adjusting sound-reinforcement systems are quantified. The impulse response, and computer-manipulated versions, can be convolved with appropriate speech and music signals for psychoacoustic testing.

2:15

Z3. The effect of oil and water on the absorption coefficient of selected acoustical materials. J. Alton Burks and E. R. Spencer (U. S. Bureau of Mines, P.O. Box 18070, Pittsburgh, PA 15236)

An acoustical material utilized in the environment of an underground mine is subject to environmental deterioration from hydraulic fluid, moisture, and dust. These and other factors can cause physical degradation of the material and can affect the material's acoustical absorptive properties. The sound absorption coefficient of 16 different acoustical materials was measured after four different sample treatments: (1) clean and dry materials as a control, (2) immersion in water and draining, (3) immersion in 100 pct petroleum-type hydraulic fluid and draining, and (4) exposure in a coal mine. The standing wave method was used to measure the normal absorption coefficient. It was found that the absorption coefficient of most materials was adversely affected by the retention of either oil or water, with oil having the greater effect. The only class of materials which was affected by neither oil nor water was neoprene foam.

2:30

Z4. Acoustical sealing at common walls. Ballard W. George (Earth Metrics, Inc., Burlingame, CA 94010)

Caulking is one of the significant elements in the achievement of effective sound insulation between adjacent spaces or dwelling units. This paper presents calculations and discussion related to the benefits of caulking, with particular reference to sealing at the base of common walls and at electrical boxes. Data in the technical literature are used to derive an estimate of the effective "sound transmission class" associated with one or more beads of caulking. Using these data, calculations are made of sound leakage transmission for various configurations of the wall base. For analysis of double-stud walls, these have been modeled as acoustical plenums, and on this basis are seen to have some advantages in terms of the leakage path attenuation. A generally similar approach is used to calculate the attenuation achievable at boxes installed in common walls.

2:45

Z5. Some thoughts on impact isolation evaluation. Edward A. Daly (Daly Engineering Company, 11855 S.W. Ridgecrest Drive, Beaverton, OR 97005) and Kerrie G. Standlee (Van Gulik/Oliver, Inc., 543 Third Street, Lake Oswego, OR 97034)

As the Portland condominium market was developing in the 70s each of the authors encountered problems using standard impact isolation test data as a design aid. There was no correspondence between subjective response received and that predicted. In response, each independently developed and used a simplified field test for evaluating modifications of ceiling/floor systems. The authors became aware of the other's work. They jointly ran a series of tests using both of the field tests methods. The findings of their joint study indicate (1) subjective response is closely related to the amount of low-frequency sound generated by impacts, (2) the impact mass used and the construction of the floor/ceiling system directly affect the level of sound generated below 250 Hz and (3) most normal floor isolation treatments are effective only in the frequency bands above 250 Hz. These findings support the following conclusions: (1) the standard test should be a field test to account for floor system construction effects, (2) the test should include frequencies below 100 Hz, and (3) the impact mass should sufficiently generate low-frequency sound.

3:00

Z6. Uncertainty evaluation for the measurement of acoustic impedance in a freefield with two microphones. Y. Champoux and A. L'Espérance (Mechanical Engineering Department, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

The measurement of the acoustic impedance and/or the reflection coefficient of an acoustical system placed at the end of an impedance tube has recently generated great interest. The development of the two-microphone method has, for one thing, greatly increased the ease and speed with which such acoustical measurements can be made. The use of a small sample size inserted in a tube is, however, still required. Recently [J. F. Allard and B. Sieben, J. Acoust. Soc. Am. 77, 1617–1618 (1985)] suggested measurement of the acoustic impedance in the freefield. The present paper addresses theoretically the accuracy of such a freefield measurement. With the hypothesis of spherical wave propagation, the uncertainties associated with the finite-approximation error, phase mismatch and the near-field error are evaluated for the case of various acoustical materials. The importance of the selection of the appropriate type of wave propagation in the calculation of the reduced impedance at the surface of the sample is also demonstrated.

WEDNESDAY AFTERNOON, 10 DECEMBER 1986 SALON 3, 1:00 TO 2:00 P.M.

Session AA. Biological Response to Vibration IV: Distinguished Lecture on the Biological Effects of Vibration on the Hand and Arm: Historical Perspective and Current Research

John Erdreich, Chairman
Ostergaard Associates, 115 Bloomfield Avenue, Caldwell, New Jersey 07006

Invited Paper

1:00

AA1. Biological effects of vibration on the hand and arm: Historical perspective and current research.
William Taylor (Netherbanks, Watten Wick, Caithness, Scotland KW1 SXJ)

Raynaud's phenomenon of occupational origin, or the *vibration syndrome of hand and arm*, was first recognized in North America by Dr. Alice Hamilton in the limestone quarries of Bedford, Indiana around 1890 to 1900. It is a tribute to the American pioneers in this area that the Acoustical Society of America now formally recognizes Biological Response to Vibration as a technical area. The objective of this paper is to highlight gaps of information regarding mechanisms of vascular, neurological, and musculo-skeletal damage caused by vibration. Also addressed is evidence that high noise level may act synergistically to the development of vibration syndrome of the hand and arm. Areas of research currently active in psychophysical and neurophysiological investigations to increase our understanding of tactile and spatial discrimination are discussed. Although the importance of sensory loss or "fine touch" is understood there is neither a proven objective scientific test with which the syndrome can be diagnosed nor is there a scale of damage assessment. Determining the exact role of the central nervous system in assessing damage from vibration is difficult in view of nonspecific symptoms reported from Eastern Europe and from Japan. To complicate matters still further, there is the possibility that repeated, rapid mechanical movements of the hand and arm associated with handling heavy tools produce Carpal Tunnel Syndrome but that the injury is not directly attributed to vibration. Therefore, it follows that there could exist an element of Carpal Tunnel Syndrome in many vibration syndrome cases. Clearly, there is an area within the framework of the Acoustical Society of America for worldwide cooperative efforts to solve many fundamental problems concerned with the effect of vibration on a hand or arm.

Session BB. Noise V, Shock and Vibration V, and Musical Acoustics IV: Demonstration Experiments to Show Fundamentals of FFT Analyzers

Frank H. Brittain, Chairman
Bechtel National, Inc., 50 Beale Street, San Francisco, California 94105

Invited Paper

2:00

BB1. Demonstration experiments to show fundamentals of FFT analyzers. Frank H. Brittain (Bechtel National, Inc., 50 Beale Street, San Francisco, CA 94105) and John Pryshepa (Bruel and Kjaer, Inc., 363-B Vintage Park Boulevard, Foster City, CA 94404)

A series of demonstration experiments will be performed to supplement the special session on FFT analyzers for noise and vibration from a user's perspective. These experiments will present some fundamentals of FFT analyzers, primarily single-channel analyzers. Fourier representation of a periodic signal will be used to illustrate the time and frequency domains. The interrelationship between frequency range, number of lines, and resolution will be shown. The concepts of sampling, folding, leakage, and windows will also be illustrated. Some noise and vibration data will be used to indicate the types of information that can be obtained using FFT analyzers. The experiments will be used to demonstrate fundamentals common to all analyzers and not any specific hardware.

WEDNESDAY AFTERNOON, 10 DECEMBER 1986 SALON 1, 1:30 TO 3:15 P.M.

Session CC. Physical Acoustics IV: Turbulence and Bubbles

Seth Putterman, Chairman
Department of Physics, UCLA, Los Angeles, California 90024

Contributed Papers

1:30

CC1. Sound amplitude fluctuations in the presence of atmospheric turbulence. Henry E. Bass, Walt McBride, and John Noble (Physical Acoustics Research Group, The University of Mississippi, University, MS 38677)

The fluctuations in received sound pressure levels at a microphone array 1 m above the ground and 5–100 m from a source 1–30 m off the ground have been recorded simultaneously with fluctuations in wind speed and temperature. The recordings were then played back through a multichannel analyzer which gives the number of acoustic cycles with peak amplitudes within 1024 amplitude intervals. Normalizing by the number of counts gives the probability of observing a given amplitude. These measurements were made at octave-band preferred frequencies between 63 Hz and 8 kHz. At frequencies of 500 Hz and below, the probability distribution appears Gaussian. At higher frequencies, the distribution is better represented by a log normal distribution. In each case, the shape of the curve was dependent upon turbulence parameters (scale and magnitude). [Work supported by the Army Research Office.]

1:45

CC2. The effect of varying scale on acoustic propagation over a smooth surface. Michael T. Bobak and Richard Raspet (U. S. Army Construction Engineering Research Laboratory, Box 4005, Champaign, IL 61820-1305)

In an earlier paper we reported on the incorporation of a measured spatially varying turbulence model into the turbulence effects theory of Clifford and Lataitis [J. Acoust. Soc. Am. Suppl. 1 79, S19 (1986)]. In the earlier paper the theory was evaluated only in the limits $L_0 \gg (L/k)^{1/2}$ or $L_0 \ll (L/k)^{1/2}$, where L_0 is the turbulence scale, k the acoustic wavenumber, and L the propagation distance. The validity of the above conditions was compromised by the varying scale of the measured turbulence model. In this paper we report on the evaluation of the complete integrals of Clifford and Lataitis [J. Acoust. Soc. Am. **73**, 1548–1550 (1983)] and discuss how these results vary from the approximate integrals.

2:00

CC3. Experimental study of acoustic radiation from a boundary layer transition region. J. C. Perraud and A. Julienne (Office National d'Etudes et de Recherches Aérospatiales, BP 72, 92322 Châtillon Cedex, France)

Wall pressure fluctuations were measured on a rigid axisymmetric body in the CEPRA 19 low-noise, anechoic wind tunnel, using flush-mounted microphones placed from the laminar region to the fully turbulent boundary layer. Microphones placed in the laminar flow region are used to detect noise radiated from the transition region, which occurs naturally, without separation, under a slightly positive pressure gradient. Cross-spectral analyses show upstream acoustic propagation in a very wide frequency band, 4–30 kHz, detected in the laminar region. A method of conditional analysis is then used to establish the sequence of events from the onset of near-harmonic instability wave packets to the generation, about 10 ms later, of turbulent spots leading to the acoustic emission. This intermittent acoustic radiation is detected in the nearfield for wind velocities ranging from 20–70 ms. Farfield detection was not achieved probably because of instrument limitations and propagation effects. [Work supported by DRET, Direction des Recherches et Etudes Techniques.]

2:15

CC4. The linear theory of the wall-jet tone. Dorothy Innes (Department of Applied Mathematics and Theoretical Physics, University of Cambridge, Silver Street, Cambridge CB3 9EN, United Kingdom)

The wall jet is one of a class of flows, of which the jet-edge tone is typical, associated with the production of discrete frequency sound. We present an incompressible, inviscid model for the wall jet in which a uniform stream emerges from a two-dimensional duct and flows along a flat wall of finite length L. A "phase-locking" criterion leads to simple scaling laws for the tone frequency and stages of operation. Specifically, if f is the frequency of the tone generated in the nth stage, $f = (U_0/2b)(4b/L)(n + \frac{3}{8})$, the sound field assumes the familiar cardiodal shape associated with a dipole source located above a semi-infinite wall. These predictions are in agreement with the experimental data of Horne *et al.* (AIAA Paper #81-2043). [Work supported by ONR.]

2:30

CC5. Transient behavior of oscillating bubbles. Andrea Prosperetti (Department of Mechanical Engineering, The Johns Hopkins University, Baltimore, MD 21218)

Research on the behavior of bubbles in sound fields has mostly been concerned with the steady regime of oscillation. However, the use of the pulsed mode in ultrasonic devices for medical applications has rendered the transient behavior of practical interest as well. In this paper several aspects of the transient regime are investigated theoretically with particular consideration given to the thermal mechanisms affecting the internal pressure in the bubble. It is shown that these mechanisms introduce a "memory" effect in the bubble response. Analytical results are obtained for low forcing, and numerical ones when large-amplitude effects are significant. Phase change effects and gas–vapor mass diffusion in the case of hot liquids are also considered. [Work supported by ONR.]

2:45

CC6. Scattering of light by a coated bubble in water near the critical and Brewster scattering angles. Phillip L. Marston and Stuart C. Billette[a] (Department of Physics, Washington State University, Pullman, WA 99164)

Microbubbles in the ocean may be coated by a thin film of surfactant since such substances can be abundant in natural waters. Such films may affect the optical and acoustical properties of bubbles. We investigated theoretical light scattering patterns for a spherical gas bubble (of radius a) coated by a film of uniform thickness h and refractive index n_f surrounded by water of refractive index $n_w = 4/3$. The patterns were computed from the partial-wave series of Aden and Kerker for ka ranging from 100–2500, where $2\pi/k$ is the optical wavelength in water. The corresponding range of a is 7.5–189 μm; h ranged from 0–3 μm and n_f was typically real and equal to 1.5. Noncoated bubbles exhibit coarse irradiance oscillations as the scattering angle θ decreases below a critical value for total reflection ($\theta_c = 82.8°$); a broad minimum in the polarized irradiance is expected near the Brewster scattering angle $\theta_B = 106.3°$ [P. L. Marston *et al.*, Appl. Sci. Res. **38**, 373–383 (1982)]. Coatings shift the coarse oscillations towards larger θ when $n_f > n_w$ in agreement with predictions of ray optics. If the irradiance near 82.8° is used to size bubbles, the effects of this shift are negligible for anticipated coating parameters. The minimum near θ_B, however, is predicted to be significantly lifted by coatings. [Work supported by NORDA and by ONR.] [a] Present address: Hughes Aircraft Co., Building E1, M/S F187, El Segundo, CA 90245.

3:00

CC7. A model of laser-induced bubble formation and collapse mechanism. Joon-Hyuk Kim, Sangbum Lee, Hyup Yang, and Ho-Young Kwak (Mechanical Engineering Department, Chung-Ang University, Seoul, Korea)

It is well known that the high power laser can produce the breakdown of liquid [M. P. Felix *et al.*, Appl. Phys. Lett. **19**, 484 (1971)]. The bubble formation and the shock wave emission after the breakdown have been observed simultaneously [W. Lauterborn and E. J. Ebeling, Appl. Phys. Lett. **31**, 663 (1977)]. Usually, the cavity concept that the bubble is empty has been employed as an initial condition for the study of bubble motion. In this study, the initial conditions of the bubble evolution due to laser irradiation, i.e., the bubble wall velocity and the pressure inside the bubble just after the bubble evolution, were obtained from the bubble formation model proposed by Kwak and Panton [J. Phys. D **18**, 647 (1985)]. Subsequent bubble evolution were calculated numerically by using the Gilmore equation in a compressible region and by using the Rayleigh equation in an incompressible region. The elapsing time from the bubble formation to the first bubble collapse and the farfield pressure signal at the first bubble collapse are in good agreement with the experimental results [W. Lauterborn (pp. 3–12) and V. S. Teslenko (pp. 30–34), both in *Cavitation and Inhomogeneities in Underwater Acoustics*, edited by W. Lauterborn (Springer, New York, 1980)]. Also, calculation results showed that shock strength and the amplitude of the pressure wave at the first collapse are strongly dependent upon the initial bubble wall velocity.

Session DD. Psychological and Physiological Acoustics IV: Masking

David A. Nelson, Chairman

Department of Otolaryngology, University of Minnesota, Minneapolis, Minnesota 55414

Contributed Papers

1:30

DD1. Forward masking by multicomponent maskers with spectral uncertainty. Donna L. Neff, Brian P. Callaghan, and Walt Jesteadt (Boys Town National Institute, 555 North 30th Street, Omaha, NE 68131)

Large amounts of simultaneous masking can be produced by small numbers of sinusoids spread over a wide frequency range, provided the frequencies of the sinusoids are changed with each stimulus presentation. In this study we examine the effect of similar stimuli in forward masking. Threshold for a 10-ms, 1000-Hz signal was measured at delays of 4, 8, 16, or 32 ms following the offset of a masker presented at 60, 70, or 80 dB SPL. The masker was a 1000-Hz sinusoid, a broadband noise, or a multicomponent complex. The complex was composed of 2–100 sinusoids that were drawn at random for each presentation from a 300–3000 Hz range, excluding frequencies within the critical band around the signal. Masking increased as the number of masker components increased, but did not exceed that produced by the sinusoid or noise. Slopes of masking were independent of component number and decreased as signal delay increased. The data have been analyzed in terms of models developed to describe masking produced by simpler stimuli. [Work supported by NIH.]

1:45

DD2. The rise and fall of forward masking. Robert P. Carlyon (MRC Institute of Hearing Research, University Park, Nottingham NG7 2RD, United Kingdom)

Three experiments investigated the forward masking of brief sinusoids by noise bursts as a function of masker duration and of masker-signal delay. Experiment 1 showed that for a 2-kHz signal and a 5-ms delay, long-duration maskers produced much more masking than brief maskers. Recovery from the longer maskers was more rapid so that at a delay of 40 ms this difference was reduced. Experiment 2 showed that, for a 2-kHz signal, the transition from simultaneous masking to forward masking was accompanied by a large drop in threshold. For a 250-Hz signal this large drop occurred as delay time was increased from 10 to 20 ms. This result was attributed to ringing on the basilar membrane. Experiment 3 compared the recovery functions for three maskers of different durations, with masker levels adjusted to produce equal masking of a 2-kHz signal at a 5-ms delay. Brief intense maskers produced slower recovery than longer weaker ones. This finding has implications for attempts to model recovery from forward masking.

2:00

DD3. The effects of a broadband masker on frequency DLs for short tones. Richard L. Freyman and Uma Balakrishnan (Department of Communication Disorders, University of Massachusetts, Amherst, MA 01003)

In an ealier study [R. L. Freyman and D. A. Nelson, J. Acoust. Soc. Am. Suppl. 1 75, S6 (1984)], it was observed that normal subjects' frequency DLs for very short tones often do not improve monotonically as the level of the standard and variable tones is increased. In some subjects (at least at 1 kHz), performance deteriorated as the test-tone level was increased from approximately 30–80 dB SPL, then improved again at still higher levels. The current experiment investigated the effects of broadband noise on the size of frequency DLs for short tones presented at moderately high levels. Frequency DLs for 5-ms, 1-kHz tones were first obtained in quiet as a function of stimulus intensity. DLs were then obtained with the signal level fixed at the SPL at which the largest DL in quiet was observed—this time in the presence of varying levels of white noise that partially masked the tones. The results indicated that changing the signal-to-noise ratio for fixed SPL tones produced similar changes in the frequency DL to those observed when the presentation level was varied in quiet. That is, in subjects with poorer DLs at moderately high levels in quiet, discrimination performance at those same levels improved when white noise was introduced. The size of the DL in noise was found to be comparable to that obtained in quiet when equated for SL. [Work supported by BRSG.]

2:15

DD4. On burst noise detectability in synthetic and natural speech. Hideki Kawahara (NTT Basic Research Laboratories. 3-9-11 Midoricho Musashino, Tokyo 180, Japan)

The relation between the hearing threshold of burst noise and some speech properties was measured by the sequential estimation method. A noise burst with a duration of 2–100 ms, LPC-based synthetic speech, and natural speech were used. Synthetic speech samples were generated from LPC parameters of the original speech samples—five Japanese vowels spoken by female speakers—with noise and pulse train excitation. Speech samples and a noise burst were added to produce stimuli, and the subjects were instructed to detect the noise burst in it. The detectability of a short noise burst (by 10 ms) was found to increase with increments of pitch duration and differ from vowel to vowel. However, the detectability of a relatively long noise burst (by 100 ms) seemed independent of excitation source parameters: The synchronization effect between short-term speech energy and minimum audible burst noise level was also observed for a very short noise burst (by 3 ms). A signal detection model of the human auditory system was proposed to confirm good correlation with these results. These results prove that it is possible to improve synthetic speech quality and waveform coding system performance.

2:30

DD5. Comodulation detection differences and comodulation masking release with multiple cue bands. Beverly A. Wright and Dennis McFadden (Department of Psychology, University of Texas, Austin, TX 78712)

Detection thresholds were determined for a noise band signal centered at 2500 Hz in the presence of four additional "cue" bands at 1500, 2000, 3000, and 3500 Hz. Contrary to the typical results of comodulation masking release (CMR), subjects were 6–12 dB more sensitive when the signal band's temporal envelope was *un*correlated with the four correlated cue bands than when all five of the bands were correlated. We refer to this difference in detectability as a comodulation detection difference (CDD). Interestingly, when none of the five bands was correlated, performance was essentially the same as when all five of the bands were correlated. The method was two-interval forced choice. The signal and the cue bands were computer synthesized and were all 100 Hz wide. Different noise samples were presented within and across trials. Results will also be presented for CDD conditions in which different subsets of the four cue bands were correlated, in which the cue bands had different center frequencies, and in which fewer cue bands were used. In addition, data will be presented from more traditional CMR conditions in which a 2500-Hz tone was masked by a noise band centered among the same cue bands used in the CDD experiments. [Work supported by NINCDS.]

2:45

DD6. Temporal effects in simultaneous masking and their influence on tuning curves in hearing-impaired listeners. Barry P. Kimberley, David A. Nelson (Department Otolaryngology, University of Minnesota, Minneapolis, MN 55414), and Sid P. Bacon (Division of Hearing and Speech Science, Vanderbilt University School of Medicine, Nashville, TN 37212)

Recent investigations raise questions about the temporal development of tuning in the auditory system [S. P. Bacon and B. C. J. Moore, J. Acoust. Soc. Am. **80**, in press (1986)]. They demonstrated that simultaneous masking psychophysical tuning curves (PTCs) were broader, on the high-frequency side, when the probe tone coincided with masker onset that when it was temporally centered in the masker. We replicated their experiment in normal-hearing listeners and extended it to hearing-impaired listeners who demonstrated abnormal PTCs. Simultaneous PTCs for 10 dB SL 20-ms probe tones, masked by 400-ms masking tones, were obtained with the probe at the onset and at the temporal center of the masker. Results from normal-hearing listeners confirmed earlier findings. However, in hearing-impaired listeners there was little change in PTC shape with temporal position. These data suggest that the mechanism responsible for the temporal effect in normal ears are ineffective in listeners with sensorineural hearing loss. [Work supported by NINCDS.]

3:00

DD7. Temporal masking curves and recovery from adaptation in hearing-impaired listeners. David A. Nelson (Department of Otolaryngology, University of Minnesota, 2630 University Ave. S. E., Minneapolis, MN 55414), and Richard L. Freyman (Department of Communication Disorders, Univ. Massachusetts, Amherst, MA 01003)

Temporal masking curves, obtained from 12 normal-hearing and 16 hearing-impaired listeners using 200-ms, 1000-Hz pure-tone maskers and 20-ms, 1000-Hz fixed-level probe tones, were fit by an exponential function with a single level-independent time constant. Normal-hearing listeners' time constants ranged between 37–67 ms, with a mean of 50 ms. Hearing-impaired listeners' time constants ranged between 58–114 ms and increased exponentially with hearing loss according to the function $\tau = 52e^{0.011(\mathrm{HL})}$, when the slope ($k$) of the growth of masking was restricted to unity. When iterative fitting procedures included k as a free parameter, k values from normal-hearing listeners varied around unity (mean = 1.03), while those from hearing-impaired listeners were less (mean = 0.71). These data indicate that temporal resolution is poorer in hearing impaired listeners by factors as large as 2.3 for a hearing loss of 52 dB, but the major effect of hearing loss on the detectability of sequential signals is due to the sensitivity loss and to an exponential recovery process that is dependent upon sensory response not stimulus level. In impaired ears, sensitivity loss apparently reduces the sensory response to high-level maskers so that the recovery process is slower, much like it is for low-level maskers in normal-hearing listeners. [Work supported by NINCDS.]

3:15

DD8. The overshoot effect in impaired and normal ears. Robert P. Carlyon and Eileen P. Sloan (MRC Institute of Hearing Research, University Park, Nottingham, NG7 2RD, United Kingdom)

The threshold for the detection of a brief tone masked by a longer duration noise burst is higher when the tone is presented shortly after the onset of the noise than at longer delay times. This finding has been termed the "overshoot" effect [E. Zwicker, J. Acoust. Soc. Am. **37**, 653–663 (1965)]. This experiment compared the size of the effect in the normal and impaired ears of five subjects with high-frequency unilateral hearing loss of sensorineural origin. Thresholds were measured for 5-ms, 4-kHz tones presented 10, 200, and 390 ms after the onset of a 400-ms, 2–8 kHz noise burst. The better ear of each subject was tested using two noise levels, one equal in sound-pressure level and one in sensation level to that used for the impaired ear. Thresholds for all subjects and all ears decreased monotonically with increasing delay time, the size of the effect being 5 dB. Thus a small overshoot effect was observed regardless of hearing impairment.

WEDNESDAY AFTERNOON, 10 DECEMBER 1986 SALON E, 1:30 TO 3:18 P.M.

Session EE. Speech Communication IV: Speech Features and Acoustical Cues

John Kingston, Chairman

Department of Modern Languages and Linguistics, Cornell University, Ithaca, New York 14853

Contributed Papers

1:30

EE1. Intraspeaker variability on nasal consonant [m]. Herbert J. Oyer, Yingyong Qi, and Claude Lambert (Speech and Hearing Science, Ohio State University, Columbus, OH 43210)

Our previous study determined that intraspeaker variability is small for the first formant of the nasal /m/ when read in a carrier sentence and larger when read in a paragraph. This study investigates factors responsible for increase in intraspeaker variability. Syllables [mi], [ma], and [mu] were each read ten times and recorded (1) in a carrier sentence with a slow rate, (2) in a carrier sentence with rapid rate, and (3) in a paragraph with normal reading rate. First formant frequency contours were compared to determine effects of context and rate on intraspeaker variability. Findings suggest that intraspeaker variability of the nasal /m/ is closely related to context, but rate of utterance is somewhat unimportant. The implication is that, although the /m/ is produced within a vocal tract that remains relatively constant, it is still subject to some variation due to the effects of coarticulation. A comprehensive study of the sounds subject to the least coarticulatory effects may prove to be important in speaker identification.

1:42

EE2. Invariant acoustic cues in stop constants: A cross language study using the Wigner distribution. H. Garudadri (Electrical Engineering, University of British Columbia, Vancouver, British Columbia, V6T 1W5, Canada), J. H. V. Gilbert, A-P. Benguerel (Audiology and Speech Sciences, University of British Columbia, Vancouver, British Columbia, V6T 1W5, Canada), and M. P. Beddoes (Electrical Engineering, University of British Columbia, Vancouver, British Columbia, V6T 1W5, Canada)

The Wigner distribution (WD) is a powerful tool in analyzing the speech signals [Garudadri *et al.*, J. Acoust. Soc. Am. Suppl. 1 **79**, S94 (1986)]. The WD is being used to investigate acoustic invariance in stop consonants. In a pilot study of CV syllables, bilabials showed an energy concentration in the low-frequency region during the burst, followed by the formant pattern of the vowel. Alveolars had a diffuse spectrum prior to the vowel region; they were also characterized by an absence of low-frequency energy during the burst of the voiceless tokens. Velars displayed a distinctly compact spectrum centered near the frequency of the second formant of the vowel. The diffuse nature of the bilabials and alveolars, reported by Lahiri *et al.* [J. Acoust. Soc. Am. **76**, 391–404 (1984)] and by the authors referenced therein, can be attributed to the temporal averaging of the burst spectrum and the diffuse vowel spectrum. This smear does not occur in velars because their burst is quite long. The WD allows one to study the burst and vowel spectra independently of each other without introducing any substantial time averaging. A formal study of the invariant acoustic properties of stop consonants from English, French, and Telugu is underway. The results of the study will be presented. [Work supported in part by NSERC, Canada.]

1:54

EE3. VOT variability: Within-subject and between-subject measurements of stop-consonant production by female talkers of English, Japanese, Navajo, and Spanish. Judith L. Lauter and Nancy B. Pearl (Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721)

Voice onset time (VOT) presents the language learner with a problem in temporal perception and generalization. However, the variability of this cue in real-speech tokens has not been reported. Two female native talkers for each of four languages—English, Japanese, Navajo, and Spanish—produced six repetitions of their native-language stop consonants in a nonsense syllable form: ha–ba, ha–da, etc. Talkers sat in an anechoic chamber 0.5 m from an AKGC451E microphone and self-monitored syllable peak levels using a sound level meter. Productions were digitized (Nakamichi DMP100 pulse-code modulator) and stored on videotape (Fisher 805A video cassette recorder). The VOTs were measured from waveforms produced on a Kay 7800 digital sonograph. Results illustrate: (1) individual differences in VOT patterns; (2) patterns of absolute-value range that are language specific; (3) less consistency in actual VOTs than suggested in the synthetic-speech literature; and (4) the need for a multicue study of phoneme perception in order to examine individually characteristic trade-offs among cues that may be the rule rather than the exception in real speech. [Work supported by AFOSR.]

2:06

EE4. Acoustic properties of geminate consonants. Aditi Lahiri (Max Planck Institut für Psycholinguistik, Wundtlaan 1, NL-6525 XD, Nijmegen, The Netherlands) and Jorge Hankamer (Linguistics, UCSC, Santa Cruz, CA 95064)

It is generally assumed in informal descriptions that what distinguishes geminate from nongeminate consonants is duration of articulation. This has not, however, been demonstrated instrumentally, nor are informal descriptions precise about what should be measured. To study the acoustic features of geminates, we investigated geminate stops in Turkish, comparing geminate and nongeminate stop consonants in similar environments. We measured the duration from the beginning of the preceding vowel to the beginning of the consonant closure, the pause duration from the end of the preceding vowel to the beginning of the following consonant, the duration of consonant closure, and the ratio of the rms energy at the onset of the consonant to that of the following vowel. The only significant difference was the duration of the pause. To investigate the perceptual salience of this feature, we took 20 minimal pairs of geminates and nongeminates, cross spliced the pause, and presented a randomized set of words to native speakers of Turkish, who wrote down what they heard. The pause was found to be a significant cue in discriminating between geminate and nongeminate consonants.

2:18

EE5. Vowels and diphthongs in Icelandic. Michel T. T. Jackson (Department of Linguistics, UCLA, 405 Hilgard Avenue, Los Angeles, CA 90024)

A factor model for midsagittal x rays of the Icelandic long and short vowels [i, I, e, y, ø, u, o, a] [M. Pétursson, Phonetica **29** (1–2), 22–79 (1974); M. Pétursson, *Les Articulations de l'Islandais à la Lumière de la Radiocinématographie* (Klincksieck, Paris, 1974)] was constructed. The factor model is based on single frames from the center of the vowels, and accounts for 95% of the variance in tongue positions with three factors. Sequences of x rays of the long and short diphthongs [ei, øi, ai, ou, au] including portions of the consonantal on- and off-glides can be described by the same factors (> 80% of the variance accounted for). We approximate the trajectories of the diphthongs by constructing curves that interpolate the measured positions in factor space. Examining the trajectory through factor space suggests an articulatory parametrization of diphthongs. [Work supported by NIH.]

2:30

EE6. Vowel features in Akan and English. Mona Lindau-Webb (Department of Linguistics, University of California, Los Angeles, CA 90024)

Height and backness are basic vowel features in most languages. Even large vowel systems, like English, can be contrasted with only these two features. When languages have additional contrasts, such as advanced tongueroot, these features are treated as if they are combined with height and backness in an additive way. The harmonizing vowels of Akan are usually described by height and back, and advanced tongueroot for the vowel harmony. Are these three features of Akan vowels phonetically the same as those in English, with an added feature of advanced tongueroot, or are they organized differently? Using PARAFAC-analysis, Harshmann *et al.* ["Factor analysis of tongue shapes, J. Acoust. Soc. Am. **62**, 693–707 (1977)] found that the tongue shapes of English vowels are controlled by two parameters. The tongue shapes of Akan vowels from four speakers were analyzed in the same way. The results show that the tongue shapes of Akan vowels are controlled by two mechanisms, which are different from those used by English speakers. Thus the "same" features in different languages are controlled differently, and when more features are required, these are not simply added, but require a different organization and integration of the controlling mechanisms. [Work supported by NIH.]

2:42

EE7. Pharyngealization in Chechen. John Kingston (Department of Modern Languages and Linguistics, Cornell University, Ithaca, NY 14853) and Johanna Nichols (Department of Slavic Languages and Literatures, University of California, Berkeley, CA 94720)

An LPC analysis was used to extract formant frequencies from stop release through vowel in eight Chechen minimal and near-minimal pairs contrasting plain versus pharyngealized syllables of the structure pa, ta, ba, da (seven tokens of each, each pronounced by two native speakers). Pharyngealization proves to raise F_1, lower F_2 (with clear sensitivity to point of articulation of the stop), and lower F_3. It thus has little observable effect on the consonant, but markedly increases the compactness of the vowel. The effect on the vowel is as predicted by Fant [*The Acoustic Theory of Speech Production* (Mouton, The Hague, 1960)], although the F_1 raising is somewhat less than would be expected. The acoustic effect of pharyngealization is similar to that of uvularization, but not identical (the degree of F_2 perturbation and the type of F_3 perturbation differ). This explains both the fact that pharyngealization can co-occur with uvulars, and the cross-linguistic rarity of this co-occurrence. It also explains the near-identical patterning of pharyngealization in Chechen and Semitic languages, and uvularization in Kartvelian languages.

2:54

EE8. The role of visual information from a talker's face in the processing of place and manner features in speech. Kerry P. Green and Patricia K. Kuhl (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

A number of studies have indicated that a dependency relation exists for the processing of place and manner features of articulation. Other studies have demonstrated that visual information provided by a talker's face can influence the perceived place of articulation of well-specified auditory tokens presented in synchrony with the visual information. Thus, when /bi/ is presented auditorially and /gi/ visually, a person perceives the talker as saying /di/. These findings raise the possibility that visual place information might also influence the processing of VOT. Members of an auditory continuum ranging from /ibi/ to /ipi/ were paired with a video display of a talker saying /igi/. The auditory tokens were heard as ranging from /ibi/ to /ipi/, while the audio/visual tokens were perceived as ranging from /idi/ to /iti/. We found that the VOT boundary for the audio/visual tokens was located at a significantly longer value than the VOT boundary for the auditory continuum presented without the visual information. These results indicate that, under certain circumstances, the processing of VOT information takes into account both auditory and visual place information. [Research supported by NIH.]

3:06

EE9. Distribution of sounds within and between Spanish syllables. Miguelina Guirao and María García Jurado (Laboratorio de Investigaciones Sensoriales, CONICET. CC 53, 1453 Buenos Aires, Argentina)

The data presented in this work are a complement to a previous statistical count [M. Guirao and M. García Jurado, J. Acoust. Soc. Am. Suppl. 1 **78**, S55 (1985)]. A corpus of 68141 syllables distributed in 159464 phonemes was used. This time our attention was oriented toward (a) distribution of sounds in each one of four syllabic positions, and (b) distribution in final position according to number of elements in the syllable. The first count indicates that fricative /s/, bursts /t k d p b/, and periodic sounds /n m l r/ cover about 80% of all items in first position. The second position in syllables of two or more components (64572 syllables) is taken mainly by vowels /e a o/. In the third position (22736 syllables) predominate /s/, /e a/, and /r n/. In the remaining syllables (3956) only /n/ and /s/ were found in the fourth position. The second count indicates that final sounds in two components syllables are vowels, 96%. With three components, /s n r/ scored 58%, /a e/ 24%, and /l/ 12%. With four, /n/ 40%, /s/ 29%, and /l r m/ 16%. With five components, /n/ 58%, /s/ 30%, and /l/ 5%. Sounds are classed according to their position and distribution within and between syllables.

WEDNESDAY AFTERNOON, 10 DECEMBER 1986 — SALON G, 1:30 TO 3:00 P.M.

WEDNESDAY PM

Session FF. Underwater Acoustics V: Propagation Modeling

Richard Evans, Chairman

ODSI, North Stonington Professional Center, Routes 2 and 184, North Stonington, Connecticut 06359

Contributed Papers

1:30

FF1. Normal mode propagation in a commercial catfish pond. Lambert E. Murray and Kenneth E. Gilbert (Institute for Technology Development, 3825 Ridgewood Road, Jackson, MS 39211 and Physical Acoustics Laboratory, University of Mississippi, University, MS 38677)

Acoustic propagation measurements were made in a commercial catfish pond in the frequency range 200–2000 Hz. A low-frequency cutoff was observed at 600 Hz. Above the cutoff frequency, modal propagation was observed. At 1000 Hz, measurements of propagation loss versus range indicated the presence of a single propagating mode. A second mode appeared at 1400 Hz and became stronger with increasing frequency. Measurements at a fixed range of the pressure field as a function of depth also indicated a single mode at 1000 Hz and showed a second mode as the frequency increased. Comparisons of the data with preliminary theoretical predictions will be presented.

1:45

FF2. Use of SALT tables for rapid calculation of sound angle, level, and travel time. Homer P. Bucker (Naval Ocean Systems Center, Code 541, San Diego, CA 92152-5000)

There are cases where a very quick calculation of sound propagation is required. One example is a real-time torpedo simulator where only a fraction of a second is available to determine sonar returns. Another example is a bistatic reverberation model where thousands of scattering elements must be included in a realistic model. A SALT table is a list of sound angle, level, and travel time listed as a function of range from the source or receiver. The number of tables used depends upon available computer storage and time available to access the data in the tables. This paper will discuss two algorithms for calculating SALT tables. In one method, the table is defined by the number of bottom reflections or ray turnunders. In the second method, the tables are numbered according to order of calculation. Several examples will be given showing reasonable agreement between calculations using SALT tables and those using more exact methods.

2:00

FF3. Long-range arrival structure using timefront analysis. Gary E. J. Bold (Physics Department, Auckland University, Private Bag, Auckland, New Zealand), Theodore G. Birdsall, and Kurt Metzger, Jr. (EECS Department, University of Michigan, Ann Arbor, MI 48109)

If the classical differential equations defining rays are rewritten in terms of travel time instead of ray path distance, it is straightforward to trace families of rays from a point with time as the independent variable. Points on rays having the same time coordinate define a timefront surface, thus designated to emphasize its high-frequency nature. Timefronts are spherical at short ranges, slowly evolving into an accordian structure. The shape of the timefront changes relatively slower over time scales of seconds, enabling a physical picture of long distance arrival times to be seen by inspection. A simple algorithm for deducing accurate travel times of eigenrays to a given receiver steps the timefront iteratively until its branches cross the receiver coordinates. Several features characteristic of long distance arrival structure are demonstrated. Numerical results from typical profiles, some including surface ducts, are presented.

2:15

FF4. Inversion of data for propagation over graveled surfaces. R. J. Lucas[a] and V. Twersky (Mathematics Department, University of Illinois, Chicago, IL 60680)

Data obtained by Medwin and D'Spain [J. Acoust. Soc. Am. 79, 657 (1986)] for low-frequency, near-grazing propagation over gravel-covered rigid planes are inverted by applying the procedure developed earlier [Lucas and Twersky, J. Acoust. Soc. Am. Suppl. 1 79, S68 (1986)] for regularly shaped protuberances. The development is based on analytical results for a point source irradiating an embossed plane, and on the uniform asymptotic representation of the coherent field as a Sommerfeld type wave system in terms of the complex error function complement Q [J. Acoust. Soc. Am. Suppl. 1 74, S122 (1983); 76, 1847 (1984)]. Initial estimates of unknown parameters are obtained by working with elementary approximations of the Q integral, and then using the complete integral for refinements and final computations. The corresponding graphical results display the primary data trends for the magnitudes of the normalized fields and for the incremental dispersion (phase velocity) versus frequency at different ranges. The procedure delineates the roles of interference and damping on the elementary wave components of the Sommerfeld type system. [a] Visiting from Department of Mathematical Sciences, Loyola University, Chicago, IL.

2:30

FF5. Finite-element analysis of marine sediment acoustics. Siamak Hassenzadeh, Yu-chiung Teng, and John T. Kuo (Aldridge Laboratory of Applied Geophysics, Columbia University, New York, NY 10027)

The propagation of acoustic waves in a shallow water environment is controlled by the dynamic response of sediments, which are generally composite (multiphase) systems, as well as the complex geological structure that results from depositional processes. To analyze the propagation of low-frequency acoustic waves in such media, the finite-element-Biot model has been developed which incorporates the Biot's theory into the finite-element method. The Biot's theory governing the dynamic behavior of fluid-filled porous materials provides an adequate description of the effects of the intrinsic properties of the sediments, and the finite-element method has the flexibility to model large-scale inhomogeneities that are commonly encountered in marine environments. The effects of such parameters as porosity, lithology, lithification, and dissipation due to the relative motion of a viscous fluid are examined numerically. It is found that the modification of acoustic waves is a result of the intrinsic properties of fluid-saturated porous sediments as well as the interactions between the fast and the slow compressional waves and energy partitioning at interfaces between dissimilar fluid-filled sediments.

2:45

FF6. 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 initial and/or boundary conditions are presented that will determine solutions of the parabolic equations so that their convolution will give the Green's function for the Helmholz equation. Examples of such transmutation representations will be presented.

WEDNESDAY AFTERNOON, 10 DECEMBER 1986 — SALON H, 1:30 TO 3:15 P.M.

Session GG. Underwater Acoustics VI: Underwater Noise

Michael Porter, Chairman
Naval Research Laboratory, Washington, DC 20375

Contributed Papers

1:30

GG1. Non-Markov noise processes. Stephanie Novak and Lyman J. Fretwell, Jr. (AT&T Bell Laboratories, WH 14A-414, Whippany Road, Whippany, NJ 07981-0903)

The simplest model noise process is Gaussian and uncorrelated. Measurements of ambient noise pressure in the ocean, however, show correlation between time samples. A stationary Gaussian random process has a correlation function that decays exponentially with time if and only if the process is Markov. Here, only stationary noise processes are considered, but the Markov assumption is relaxed. A general algorithm for constructing a non-Markov stationary time series is described. It is assumed that the statistics are Gaussian; however, the method can also be extended to include non-Gaussian statistics. The general algorithm is then applied to a special case. A model non-Markov noise correlation function consisting of a linear combination of two exponentials with two different correlation time constants is fit to an experimental correlation function. Time series for uncorrelated, Markov and non-Markov noise processes are generated from the same sequence of random numbers and compared. Comparisons are made with observed time series. Consequences regarding signal detection are discussed.

1:45

GG2. Nonlinear and non-Gaussian ocean noise. Patrick L. Brockett, Melvin Hinich, and Gary R. Wilson (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

Bispectral analysis is a statistical tool for detecting and identifying a nonlinear stochastic signal generating mechanism from data containing its output. Bispectral analysis can also be employed to investigate whether the observed data record is consistent with the hypothesis that the underlying stochastic process has Gaussian distribution. From estimates of bispectra of several records of ambient acoustic ocean noise, a newly developed statistical method for testing whether the noise had a Gaussian distribution, and whether it contains evidence of nonlinearity in the underlying mechanisms generating the observed noise is applied. Seven acoustic records from three environments are examined: the Atlantic south of Bermuda, the northeast Pacific, and the Indian Ocean. The collection of time series represents both ambient acoustic noise (no local shipping) and noise dominated by local shipping. The three ambient records appeared to be both linear and Gaussian processes when examined over a period on the order of a minute, but were found to be nonlinear and non-Gaussian when examined over shorter time periods on the order of a second. In each case, the time series dominated by local shipping noise tested to be nonlinear and non-Gaussian over both short and long time periods.

2:00

GG. 3. Contribution of sources over slopes to midocean ambient noise vertical pattern. Donald A. Murphy and Mona J. Authement (Naval Ocean Research and Development Activity, Ocean Acoustics Division, Code 244, NSTL, MS 39529-5004)

The shape and depths of notches in the midocean vertical ambient noise patterns are heavily affected by arrivals from sources over continental slopes [D. A. Murphy and D. R. Del Balzo, J. Acoust. Soc. Am. Suppl. 1 77, S3 (1985)]. The steep angle and bottom interacting parabolic equation program PAREQ was used to simulate the field at the Contrack VI site in the Northeast Pacific due to sources over the slope. The assumed depths and distances down the slope of the noise sources were varied and the contributions were added to those of previously calculated sources to calculate the ambient noise field. For noise sources at short distances down the slope, the sensitivity to noise source depth was not significant. The distance of the noise source down the slope was important in calculating the shape of the vertical noise notches.

2:15

GG4. Sound radiated into the Arctic Ocean by cracks in the polar ice sheet. A. J. Langley (5-007 Department of Ocean Engineering, MIT, Cambridge, MA 02139)

Ambient noise in the Arctic Ocean is caused partly by cracking of the ice sheet. Starting with a suitable "equivalent source" for fracturing processes, an analytical description of the radiated field has been derived when the ice sheet is modeled by an infinite, thick, elastic plate bounding a homogeneous fluid half-space. This simple model can be used to assess the importance of radiation damping, absorption, scattering, and ocean sound-speed profile in the production of real sound fields. Under mid-Arctic conditions at frequencies below about 20 Hz, the model indicates that radiation damping and absorption are very weak, and that a large contribution to the ambient noise could come from elastic waves in the ice that are scattered by inhomogeneities or edges of floes. Therefore, models relating low-frequency ambient noise to sources in the ice should take some account of scattering. At higher frequencies, absorption becomes more important, and the effect of scattering becomes harder to estimate because it depends upon details of the ice structure. [Work supported by Arctic Program, ONR.]

2:30

GG5. Waterborne and seismic partitioning of distributed noise in shallow water. W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375) and Henrik Schmidt (SACLANT ASW Centre, I-19026 La Spezia, Italy)

The waveguide nature of a shallow water environment bounded below by a viscoelastic medium permits noise to couple into seismic waves. Geophone and hydrophone measurements [T. Akal *et al.*, in *Ocean Seismo-Acoustics*, edited by T. Akal and J. Berkson (Plenum, New York, 1986), pp. 767–778] have shown that below a threshold frequency of about 10 Hz in 100 m of water, there is an increase in the vertical and radial components of the outputs of geophones as compared to no increase in the pressure output from an adjacent hydrophone. A previously developed wave theory of distributed noise in a stratified ocean [W. A. Kuperman and F. Ingenito, J. Acoust. Soc. Am. 67, 1988–1996 (1980)] used with the appropriate viscoelastic medium Green's function [H. Schmidt and F. B. Jensen, J. Acoust. Soc. Am. 77, 813–825 (1985)] explains this phenomena which occurs below the waterborne propagation cutoff frequency.

2:45

GG6. Directional underwater noise estimation. R. W. Bannister (Defence Science Establishment, Auckland, New Zealand), A. S. Burgess (Weapons Systems Research Laboratory, DSTO, Department of Defence, GPO Box 2151, Adelaide, SA, Australia), and D. J. Kewley (Naval Underwater Systems Center, New London, CT 06320)

In the waters surrounding Australia and New Zealand, there are features of the measured three-dimensional ambient noise field (20–200 Hz) which appear similar to those found in the northern hemisphere where shipping noise is dominant. In the latter case, this shipping noise can be enhanced when vessels are at high latitudes or over sloping bathymetry. However, in the southern hemisphere, there are fewer ships, negligible numbers over suitable bathymetric slopes, and practically none at all at high latitudes. This apparent inconsistency in interpretation of the broad features between the different hemispheric noise data had led to the development of the directional underwater noise estimates (DUNES) model for ambient noise. The model includes wind-generated noise from local, bottom bounce, RSR, and RR path sources. The last sources include the high latitude winds and continental or seamount slope enhanced wind noise [R. W. Bannister, J. Acoust. Soc. Am. 79, 41–48 (1986)]. The same wind-generated surface source levels {based upon upgraded values [A. S. Burgess and D. J. Kewley, J. Acoust. Soc. Am. 73, 201–210 (1983)]} are used for each source location and when combined with an appropriate transmission process produces predictions of vertical noise directionality and resulting omnidirectional noise. Features included are a speed and frequency-dependent wind-source depth, season, latitude, depth, and bottom reflection loss dependences, and specific shipping sources, with or without slope enhancements. Comparison with vertical directionality data shows that the levels and frequency relations of the various angular components match well.

3:00

GG7. High sea state ambient noise near Bermuda. A. Donn Cobb, William A. VonWinkle, and David G. Browning (Naval Underwater Systems Center, New London, CT 06320)

The observed relationship between wind speed and ambient noise level has varied from ocean to ocean. Generally higher levels measured in southern hemisphere oceans have been attributed to fully developed seas, although this mechanism is not understood. Ambient noise data taken near Bermuda after a persistent (5 days) wind of approximately 40 kn are higher than previously observed and consistent with southern ocean measurements, thus supporting a developed sea state hypothesis. [Work supported by NUSC.]

WEDNESDAY AFTERNOON, 10 DECEMBER 1986 — SALONS E AND F, 3:30 TO 5:00 P.M.

Plenary Session

Ira Dyer, Chairman
President, Acoustical Society of America

Business Meeting

Presentation of Awards

Silver Medal in Engineering Acoustics to Albert G. Bodine
Silver Medal in Musical Acoustics to John G. Backus
Silver Medal in Noise to Tony F. W. Embleton
Distinguished Service Citation to Stanley L. Ehrlich
Distinguished Service Citation to Samuel F. Lybarger

Musical Entertainment

Musicians from the Orange County Chamber Orchestra, M. Levy, Conductor, will perform chamber transcriptions of two movements from Vivaldi's The Four Seasons, for harpsichord, cello, and violin.

THURSDAY MORNING, 11 DECEMBER 1986 — SALON K, 9:30 A.M. TO 12:00 NOON

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.

Session HH. Noise VI and Architectural Acoustics V: Outdoor-to-Indoor Sound Transmission, Part 1: FAR Part 150 Airport Programs

Dwight E. Bishop, Chairman
BBN Laboratories, Inc., P.O. Box 633, Canoga Park, California 91304-0633

Invited Papers

9:00

HH1. Sound insulation for buildings under Part 150. Richard N. Tedrick (Federal Aviation Administration, Washington, DC 25091)

Part 150 of the Federal Aviation Regulations provides a mechanism for the operators of noise-impacted airports to develop comprehensive programs to limit or reduce noise effects, including noncompatible land uses. In developing these programs, the airport operator is required to consult both with the users of the airport and with local, state, and federal agencies having land-use planning authority. These groups then examine various alternative operational and land-use actions and devise a plan suited to local conditions. One option for decreasing noncompatible uses is sound insulation of buildings in impacted areas. In fact, Part 150 itself cites several combinations of uses and aircraft sound levels that require specified noise level reductions to be considered compatible. Obviously sound insulation can be installed as a part of the original building design; however, in many cases, additional insulation may be required. Oftentimes, this supplemental insulation is eligible for funding under the federally funded Airport Improvement Program. Of the 31 noise compatibility programs that have been submitted to date, 20 have included planned increases in sound insulation in one or more structures.

9:25

HH2. Field manual for sound proofing of buildings in the vicinity of airports. Simone L. Yaniv (National Engineering Laboratory, National Bureau of Standards, Gaithersburg, MD 20877)

This paper summarizes the manual prepared for the FAA to provide uniform national guidance for soundproofing existing buildings around airports in accordance with the goals set forth in FAR Part 150. The guidelines contained in the manual are quantitative rather than qualitative in scope. Procedures are given for estimating the noise reduction provided by existing buildings in the vicinity of airports, determining the amount of soundproofing required to achieve an adequate indoor noise environment, and for selecting cost-effective retrofit options. Although the manual is intended for use by technical people without specific training in acoustics, computations are required. Accordingly, examples will be presented to illustrate the step-by-step procedures, lookup tables, and graphs used.

9:50

HH3. Basic problems in designing to control noise intrusion. J. D. Quirt (Acoustics Section, Institute for Research in Construction, National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada)

Intrusion of outdoor noise into buildings interferes with many activities. This activity interference can be controlled by proper design and construction of the building's exterior envelope. Specifying that all components should have sound transmission class (STC) ratings equal to the desired difference between outdoor and indoor A-weighted sound levels would have an appealing simplicity, but has several shortcomings. These are detailed in the course of presenting a straightforward procedure for determining suitable STC values for the components of the exterior envelope. Particular attention is given to dependence of the noise reduction on characteristics of the exterior sound field and the sound transmission properties of the building envelope.

10:15

HH4. Design and field experience in the sound insulation of dwellings around airports. David Brown (Wyle Laboratories, 128 Maryland Street, El Segundo, CA 90245)

Remedial sound insulation is currently being considered, and in some cases being applied, as an aircraft noise mitigation option for residential land uses around airports. Such large-scale programs incur very large expenditures of funds and therefore need to be carefully assessed in terms of both objective and subjective merits. This paper reviews two separate programs applied to dwellings around Los Angeles International Airport and their implications for projects at other airports. Each of the topic programs was applied to a sample of dwellings representative of typical construction and different noise exposure conditions around the airport.

The more recent program, completed in 1985, is emphasized, as it provides an insight to current noise conditions within dwellings near airports. Specific case histories are illustrated in terms of design features; technical, aesthetic, and subjective acoustical merits; and cost implications. The paper also addresses acoustical design theory in its application to this particular type of problem area.

10:40

HH5. Logan International Airport sound isulation program. Ardis Stiffler (MASSPORT, Logan International Airport, East Boston, MA 02128)

In 1985, MASSPORT completed a residential sound insulation pilot project in four homes in East Boston and Winthrop. The homes were of constuction typical of their neighborhoods and exposed to various types of aircraft noise. The acoustical treatments were successful in reducing the interior levels of noise. In October 1985, the MASSPORT Board directed that the residential sound insulation project continue with acoustical work on 150 more houses. By December 1986, construction modifications of a number of these houses should be completed. The presentation will cover the administrative organization and the acoustical requirements of the program, and will describe the procedures and approach of such a program from both the airport and the community point of view. Information will also be presented on the acoustical testing methods used by BBN Laboratories, acoustical consultants for the program, in the field before and after construction modifications.

Contributed Papers

11:05

HH6. Exterior noise spectra and interior noise. Alexander Segal (Department of Planning and Land Use, 5201 Ruffin Road, Suite B, San Diego, CA 92123-1666)

A statistical analysis was performed in order to determine the impact of different published exterior traffic noise spectral envelopes upon interior noise levels. It was found that the same exterior A-weighted sound level attributable to different traffic noise spectra produces interior sound levels ranging within 3 or more dB. The difference in interior noise levels is significantly affected by the building envelope element combinations. The effect of the relative size of building elements is less pronounced. Several traffic noise spectra were generated by computer for typical traffic speeds and mixes. Their application allowed significantly reduced variability of the interior noise results. Taking into account numerous interior noise analysis, currently performed in order to satisfy interior noise regulations, some unification of the exterior noise spectra information appears to be urgent.

11:20

HH7. An exterior enclosure sound level reduction (SLR) worksheet for use by architects and housing developers utilizing the external wall rating (EWR) method for aircraft noise. Dana S. Hougland (David L. Adams Associates, Inc., Denver, CO 80211)

A worksheet calculation method was developed for architects and housing developers to calculate the exterior enclosure SLR of new residential units. The worksheet calculations are designed to be completed and submitted to the local building department as preliminary verification that new residential construction around a major commerical airport provides the required enclosure SLRs. The worksheet is based on the external wall rating (EWR) for aircraft noise [Sharp *et al.*, "The Assessment of Noise Alternative Measures for External Noise," Wyle Res. Rep. WR76-3 (April 1976)]. The SLR worksheet and a data base of regional constructions are incorporated into proposed revisions to the city/county building code. The results of SLR measurements made on a typical, new residence in the affected area are compared with calculated data on the same residence.

11:35

HH8. Use of the Shell Isolation Rating method in the design of an elementary school near a military air base. James A. Johnson (The Joiner-Rose Group, Inc., 4125 Centurion Way, Dallas, TX 75244)

The National Bureau of Standards document NBS BSS 84, "Design Guide for Reducing Transportation Noise in and Around Buildings," was adapted for use in evaluating several design options by an architect in the course of replacing a highly noise-impacted elementary school with a new facility on the same block of land. The dominant noise sources were military fighter aircraft operating in a traffic pattern which placed them approximately 1550 ft abeam the school, some 2500 ft from the runway threshold. The predicted Shell Isolation Ratings are compared to field-measured results at the school site in order to evaluate the method's utility as a design tool. Faculty response to the new facility is favorable, partly because a noise level reduction greater than that recommended by the military was found advisable, and was achieved. Computerization of the methodology will be presented.

11:50

HH9. The coincidence dip in window sound transmission. Mark E. Schaffer (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

Glazing selections for exterior wall constructions are usually made on the basis of 1/3-octave-band sound transmission loss as measured in a dual reverberation room test facility. This selection method is generally acceptable where the exterior sound is broadband in nature. Under these conditions the coincidence effect, which is responsible for a drop in transmission loss performance, is blurred over upper frequencies and is barely noticeable to a building occupant. However, when the exterior noise contains significant amounts of sound energy in the coincidence frequency range and the sound is incident over a varying angle (e.g., an aircraft flyby or a heavy truck driveby), the coincidence effect causes the glass to act as a sweeping narrow bandpass filter which can be most annoying to a building occupant. Results of field tests will be shown to verify the angular dependency and the strength of the effect in narrow bandwidths. Audio recordings will be played to demonstrate the relationship between the coincidence effect and the angle of incidence of direct field noise.

Session II. Engineering Acoustics III: New Designs and Analyses

Alan D. Stuart, Chairman

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

Contributed Papers

9:00

II1. Transducer modeling using axial profile fitting techniques. David Bennink and Anna L. Mielnicka-Pate (Department of Engineering Science and Mechanics, Iowa State University, Ames, IA 50011)

Axial profiles of three piezoelectric, nonfocused, immersion pulse-echo transducers were measured using a small spherical reflector and an automatic scanning device. The scanning distance range included the first and second maxima. Measured data were compared with rigid disk radiation theory. The frequency-dependent active radius was used as a parameter in the curve fitting. In addition, three different approaches were used in the curve fitting. These included (1) fitting the entire range of the scanning distances, (2) fitting the first maximum, and (3) fitting the second maximum. The results of fitting the first maximum agree extremely well with the entire curve fitting, while the results obtained from the second maximum contain more errors. The results of modeling the transducers will be discussed in terms of the active radii and spatial sensitivity functions. In general, good agreement between the theoretical and experimental results was obtained. [Work supported by Center for NDE, Ames, Iowa.]

9:15

II2. TANDEL measurement of hydroacoustical particle motion. G. C. Alexandrakis[a] (Physics Department University of Miami, Miami, FL), K. M. Rittenmyer, and P. S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P.O. Box 568337, Orlando, FL 32856-8337)

A new ferroelectric sensor for measuring the particle motion in hydroacoustic fields has been developed. The sensor is a capacitor constructed by adding conducting end plates to single crystals or polymer composites of antimony sulfur iodide (SbSI). The device is operated above the Curie point of about 20 °C. The heating is accomplished through the hysteresis cycle driven by a Sawyer–Tower-type bridge. In the temperature region slightly above the Curie point, both the real part ϵ' and the imaginary part ϵ'' of the dielectric permittivity have a large negative slope as a function of temperature. The negative slope in ϵ'' causes the *T*emperature *A*utostabiling feature of the *N*onlinear *D*electric *EL*ement (TANDEL). Cooling the sensor by a fluid flow changes the temperature of the sensor, and consequently ϵ' and ϵ''. The change of ϵ' unbalances the bridge, producing a detectable signal. The output for steady flow shows the characteristic exponential dependence on fluid speed. The sensor displays directional sensitivity. The response to low-frequency acoustic fields is discussed and related to the heat transfer properties of horizontal cylinders. [Work supported by the ONR.] [a] Dr. Alexandrakis was appointed under the CNR Summer Faculty Research Program of the American Society for Engineering Education.

9:30

II3. An acoustic sensor utilizing a twisted birefringent fiber. Andong Hu and Frank W. Cuomo (Department of Physics, University of Rhode Island, Kingston, RI 02881-0817 and NUSC, Newport, RI 02841-5047)

The results of the light propagation in the linear twisted birefringent fiber using the parallel method have been obtained [H. G. Winful, Appl. Phys. Lett. **47**, 213–215 (1985)]. A conceptual design of an acoustic optical fiber sensor and the numerical calculations are given based on the amount of twist about the fiber axis generated at the output fiber end by an acoustic pressure field. If the input power, coupled into the birefringent fiber at 45°, is assumed to be 0.5 mW, and the induced output optical power is assumed to be 1.62×10^{-9} W under the twist rate condition of 1 second of arc, the minimum detectable pressure is on the order of 26 dB *re*: 1 μPa/$\sqrt{\mathrm{Hz}}$ and the approximate modulation index is 3×10^{-3}/Pa. The active area of the probe can be on the order of 1 mm and the predicted frequency response is quite broad. These results are favorable in comparison with other optical fiber pressure sensors proposed in the literature.

9:45

II4. A simplified math model and equivalent circuit for large, uniform, multilayered gratings of encapsulated compliant tubes irradiated by plane sound waves in water. G. A. Brigham (Aquasonics, Inc., Anaheim, CA 92806) and Barry Woolley (Raytheon Corporation, Portsmouth, RI 02871)

Detailed mathematical studies of encapsulated layers of compliant tubes indicate that the elastomeric effects can be reduced to a shear viscous damping and a shear wave stiffness loading in the layer equivalent circuit, which has been shown in earlier research to be a *T* network. Analysis of multiple layers yields the observation that the unsymmetric tube/layer modes are strongly coupled to the difference of the pressure across the layer while the symmetric (compliant) modes are strongly coupled to the average of the pressure around each tube. A circuit interpretation reveals all symmetric modes in parallel with their net impedance, a shunt by the main branch which consists of the antisymmetric modes in parallel, all in a common *T* network. The theoretical derivations, coupling impedances, equivalent circuit, and comparison with both data and a large computer model are all presented in this paper.

10:00

II5. Decomposing one-dimensional acoustic pressure response into propagating and standing waves. Charles E. Spiekermann (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910) and Clark J. Radcliffe (Department of Mechanical Engineering, Michigan State University, East Lansing, MI 48824)

Few actual sound fields are representative of ideal acoustic pressure responses and ideal boundary conditions, such as those found in anechoic or reverberant rooms. Normally encountered enclosures have complicated responses that are difficult to relate to a boundary condition that is in between these two ideal extremes. Yet, the propagating and standing wave responses associated with absorptive and reflective boundary conditions seen in the ideal cases are fundamental bases to understand these more complicated problems. An analytical method is developed to decompose a one-dimensional acoustic pressure response associated with a specified partially absorptive boundary condition into an equivalent summation of propagating and standing waves usually associated with absorptive and reflective boundary conditions, respectively. The propagating and standing wave responses are scaled and shifted in phase by factors that are dependent on the boundary absorptivity and frequency, but are independent of the spatial location. The complicated mixed response is decomposed into varying amounts of the ideal responses, which can be helpful during a design analysis.

10:15

II6. Stripping the one-dimensional acoustic pressure response into propagating and standing wave components. Charles E. Spiekermann (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27695-7910) and Clark J. Radcliffe (Department of Mechanical Engineering, Michigan State University, East Lansing, MI 48824)

Most realistic boundary conditions are partially absorptive, with the rest of the incident acoustic power being reflected. This results in a response that can be viewed as a summation of propagating and standing wave effects. A one-dimensional, measurement-based, two-microphone spectral analysis technique is developed which *S*eparates the *T*otal acoustic *R*esponse *I*nto *P*ropagating and *S*tanding (STRIPS) wave components. STRIPS uses measurements that contain implicit information about the boundary condition to extract the propagating and standing wave components. These sum to be the total system response without requiring explicit knowledge of the particular boundary condition. An exact decomposition example for a one-dimensional acoustic pressure model with a known mixed boundary condition is used to validate the STRIPS method. The utility of the STRIPS method is demonstrated using experimental measurements from a tube as input to STRIPS to obtain the propagating and standing wave pressure components. Changes in experimental boundary absorptivity are seen in the STRIPS results when a flared end is attached to the tube.

10:30

II7. The performance of an optical fiber pressure probe for ultrasonic studies. Frank W. Cuomo and Andong Hu (Department of Physics, University of Rhode Island, Kingston, RI 02881-0817 and NUSC, Newport, RI 02841-5047)

It has been suggested that the three-dimensional quantitative characterization of ultrasonic fields be determined utilizing a miniature optical fiber pressure probe [F. W. Cuomo, 12th International Congress on Acoustics (ICA), Toronto, Canada, paper G7-3 (1986)]. This approach has been found useful in extending experimentally the two-dimensional quantitative information now available with Schlieren imaging methods. In this paper, the experimental results of the ultrasonic optical fiber sensor related to pressure sensitivity, frequency response, and directionality are presented. The active area of this probe is less than 1 mm and the frequency range of interest lies between 500 kHz and 2 MHz.

10:45

II8. Rapid, electronic measurement system for determination of dc flow resistances. Michael R. Stinson and G. A. Daigle (Division of Physics, National Research Council, Ottawa K1A 0R6, Canada)

A measurement system will be described which allows dc flow resistances (1 to 10^4 cgs ohms) of various samples to be determined quickly and accurately. Variable-capacitance pressure transducers are used to measure pressure differences across both the test sample and a laminar flow element with known resistance (these being in series), so that the unknown resistance can be calculated directly. The rate of flow of air through the sample is adjusted using a mass flow controller; flows of 10^{-3}–1 cm^3/s can be contolled to better than 1%. The use of these electronic sensors permits measurements of flow resistance to be made much faster than with the traditional Leonard device. Initial measurements of flow resistance (i.e., the zero frequency limit of acoustic resistance) have been made on small, circular orifices in thin plates.

11:00

II9. An acoustic scoring system for a helicopter gunnery range. Robert F. Davey (Specialty Sensor Systems, 165 Spinks Canyon Road, Duarte, CA 91010)

A system has been developed which measures the position of the impact of munitions fired from helicopters by determining the arrival time of the detonation sound. The signals from acoustic transducers are digitized and then processed by a detection algorithm to determine the arrival time of the leading edge of the acoustic signature. The difference in the arrival times at three transducers establishes the location of the sound source as the intersection of two hyperbolas. Field tests of a prototype system were conducted to identify the most desirable acoustic transducer, to examine the effect of propagation through dense vegetation on the detonation signature, and to verify the accuracy of the computational algorithms. The results showed that the signals from low-frequency speakers can be detected more reliably than those from midrange driver/horn systems. With properly selected speakers sited in a triangular array, the detonation position was established to an accuracy of about 2 m. An analysis of the system performance shows that such techniques can be extended to score impacts over an area as large as 1 km^2.

11:15

II10. Paging system design concepts. Areg Gharabegian (Engineering-Science, 75 North Fair Oaks, Pasadena, CA 91109)

A new in-plant paging and emergency evacuation alert system was designed for indoor and outdoor areas of a nuclear power generating station and all support areas. The new system was designed to provide intelligible message transmission during all modes of plant operation and for day-to-day paging and emergency announcements. A comprehensive microcomputer-based acoustic analysis using Lotus 1-2-3 spread sheet software was performed to ensure intelligible speech coverage. The articulation index procedure stated in ANSI Standard S3.5 1969 was the basis for this analysis. Gated time mode measurements were conducted for the complex areas to evaluate direct-to-reverberant ratios because use of the measured reverberation time for the acoustic calculations produced unrealistic results. Based on the results of these measurements, the critical radius was calculated for different types of speakers in various areas. To provide an intelligible speech coverage, a total of 4335 new speakers, 4058 for indoor use and 277 for outdoor use, were designed. Nine different types of speakers with different power levels and directivity were utilized for this system.

Session JJ. Musical Acoustics V and Education in Acoustics: Music and Acoustics for the Millions

Dean Ayers, Chairman

Department of Physics and Astronomy, California State University, 1250 Bellflower Boulevard, Long Beach, California 90840

Invited Papers

8:30

JJ1. Radio as a forum for acoustics. Jim Metzner (Sounds of Science, Lincoln Road, Croton-On-Hudson, NY 10520)

In the context of contemporary electronic media, particularly radio, what are the challenges which face both scientist and broadcaster in presenting a scientific subject—specifically in the area of acoustics—to a general audience? The aim is to inform simply and accurately, without trivializing the subject, while taking into account that the listening audience may have little—if any—background in the field being discussed. Examples will be presented from "The Sounds of Science," an award-winning short-format daily radio series produced by the author for The DuPont Company, and distributed to over 100 public and commercial radio stations nationwide. The series uses a blend of interview and ambient sound to convey an experience of scientific research and ideas. Subjects treated in the domain of acoustics include early woodwind instruments, the sound of a space launch, Tibetan chant, and sound paradoxes. The preparation and conditions for interviews, which help to establish an exchange meaningful to both parties, will be discussed. A collaboration is fostered in which the scientist often plays an active role in how best to portray his or her research in sound.

9:00

JJ2. Sound through music: Exhibit-based teaching at the Exploratorium. Thomas Humphrey (The Exploratorium, 3601 Lyon, San Francisco, CA 94123)

Acoustical topics taught at the Exploratorium include speech, hearing, language, psychoacoustics, and the physics of sound. We have found that each of these topics can be reached easily by beginning with a discussion of music. The evolution of the western scale, musical composition, and the design of musical instruments involve the concepts of a vibrating string and its harmonics, logarithmic hearing, acoustic resonance, and physical beats. The broadness of this approach leads naturally to discussions of the bases of aesthetics, the ability of the sciences to contribute to an appreciation of the arts, and the natural creative relationship between artists and scientists. The exhibits used in this exploration are all hands-on, allowing for a variety of approaches to teaching. The students use the exhibits both on their own and with the guidance of an Exploratorium teacher. Independent use allows for self-discovery, while the guide can provide direction toward less obvious observations. Classroom discussions typically follow the floor work.

9:30

JJ3. Educating Americans for the 21st century. Francis P. Collea (Mathematics and Science Advisor, Academic Affairs, Office of the Chancellor, California State University, 400 Golden Shore, Long Beach, CA 90802)

Recent national commission reports have discussed a new era for science education. Unlike the sciences of the 1950s and 1960s, this new era has raised some provocative questions concerning science education for the future. The science movements of the post-sputnik period stressed science for the elite, more science and mathematics for the college-bound student, developing a corps of the very best engineers and scientists who could take us to the moon and let us assume the leadership in the international scientific enterprise. It was a time whose purpose was to get the most able students from our high schools into the calculus and physics courses at universities and colleges as quickly as possible. The recent report of the National Science Board's Commission on Precollege Education in Mathematics, Science, and Technology represents a radical departure from the movements of the 50s and 60s. The purpose of this paper is to discuss the implications of their recommendations for the future of science education.

10:00

JJ4. The CSULB Mobile Science Museum: Sights and sounds of science. Michael S. Schaadt (School of Natural Sciences, California State University at Long Beach, Long Beach, CA 90840)

In 1980, a 27-ft recreational vehicle was adapted to carry interactive science displays to local schools and community groups. This Mobile Science Museum (MSM) exhibits as many as 40 individual hands-on displays both inside as well as on tables directly outside the vehicle. University science students serve as docents and pass on their enthusiasm for science while providing role models for young visitors. Most displays are borrowed from university science teaching and research laboratories, while others are developed and fabricated by faculty, staff, and students. Topics found to be particularly effective for presentation in the hands-on mode include sound, light, and marine biology. Far more schools request visits by the MSM than can be served within current budgetary constraints. While this project is supported by CSULB, School of Natural Sciences, the majority of the operating budget comes from other sources of funding, including companies specializing in science and technology, school districts, parent/teacher associations, and private individuals.

Contributed Papers

10:30

JJ5. Inexpensive microcomputer-based sound measurement apparatus. Roger J. Hanson and Robert T. Ward (Department of Physics, University of Northern Iowa, Cedar Falls, IA 50614-0150)

Hardware and software kits (less than $350) have been developed which together with an Apple II computer can be used to make a variety of sound measurements suitable for educational purposes at the precollege level. Descriptions and demonstrations will be given of such systems including those developed by TERC (Technical Education Research Centers, Cambridge, MA) for junior high school science programs. The systems include amplifiers, A to D and D to A converters, and frequency analysis components all packaged in the form of plug-in cards or modules connected to the game port. A microphone and a speaker are also included. When used with the included software, the systems can perform the elementary functions of a digital oscilloscope, a sound spectrograph, and a frequency counter. Laboratory tests of the response of the systems to various signals will be reported.

10:45

JJ6. Educational applications of microcomputer-based sound apparatus. Roger J. Hanson and Robert T. Ward (Department of Physics, University of Northern Iowa, Cedar Falls, IA 50614-0150)

Some educational uses of the equipment described in the previous paper [R. J. Hanson and R. T. Ward, J. Acoust. Soc. Am. Suppl. 1 **80**, S72 (1986)] will be described and demonstrated. It can be used to study waveforms for different sounds, measure frequencies, study voice prints (simple sound spectrograms), generate various types of sounds, and measure the speed of sound in air. Experiences of its use with junior high school students will be reported. Suggestions for use with other audiences will be given.

11:00

JJ7. Computer microworlds for sound and music. Gerald J. Balzano and Mark Dolson (Center for Music Experiment, Q-037, University of California, San Diego, CA 92093)

Computers are increasingly being used both for education in acoustics and for making music. But frequently the computer is used merely to automate an established paradigm. Thus we have the spectacle of music students using the computer as a drill-and-practice machine, of musicians using the computer as a sophisticated tape recorder, and of students in acoustics using the computer as a data-acquisition-and-plotting tool in laboratory experiments. In a more innovative "microworld" approach, the computer *is* the laboratory. In the *SESAME* project, we are designing a set of Structured Environments for Sound And Music Exploration in which students and professionals alike can use the computer to design their own musical and/or acoustical experiments. But this is feasible only with software that shields users from the machine-oriented details of conventional programming languages, allowing users to think and program directly in terms of sonic and musical structures. During the past year at U. C. San Diego we have begun working toward this goal, using the IBM-PC to control both synthesizers and digital-to-analog converters. In this report, we present some preliminary results of several examples of computer environments for sound and music appropriate for both learning and research. [Work supported by the Department of Education, Fund for the Improvement of Post-Secondary Education.]

11:15

JJ8. Laboratory and demonstration experiments with guitars. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Several interesting demonstration and laboratory experiments can be done with acoustic and electric guitars. Some of these deal with vibrations of the strings, some with the resonances of the body, and some with sound radiation from the guitar. Doing physics experiments with musical instruments increases student interest.

11:30

JJ9. Hands-on musical acoustics in the elementary classroom. R. Dean Ayers (Department of Physics and Astronomy, California State University, 1250 Bellflower Boulevard, Long Beach, CA 90840)

The author has developed a minicourse for school teachers on how they can include acoustics in their teaching of science, with particular emphasis on musical sounds. The organizing theme is the direct, hands-on accessibility of many basic ideas in this area. The course is divided into three units: (1) simple vibrators, covering ideas about pitch, frequency, mass, and stiffness; (2) vibrators with several natural frequencies, presenting concepts of standing waves, nodes, and techniques for manipulating the sounds from such a vibrator; and (3) sound waves themselves, with emphasis on the wind instruments. An important part of each unit is the "dessert"—a simple musical toy whose operation illustrates the ideas discussed: (1) a seven-note, pentatonic thumb harp (African sansa or kalimba) constructed from materials indigenous to the American shopping center; (2) a *tromba marina* (bowed monochord), also built from cheap, readily available materials; and (3) a "flugle"—a closed/open cylindrical whistle on which bugle calls can be played. A color code for pitch makes it easy for beginners to play familiar tunes.

11:45
Discussion

An open meeting for discussion and demonstrations will begin immediately following the last paper in Session JJ. This meeting will be held in Salons A and B as well as outside the hotel in the CSULB Mobile Science Museum.

Session KK. Physical Acoustics V: Scattering

Steven R. Baker, Chairman
Physics Department, Naval Postgraduate School, Monterey, California 93943

Contributed Papers

9:00

KK1. Shape characterization from the complex poles of the acoustic scattering amplitude. G. C. Gaunaurd (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20903-5000)

It was found that every penetrable scatterer has a peculiar set of poles in the lower half of the complex-frequency plane ($x = ka$). Some of these poles depend only on shape while others depend only on composition. The poles are labeled by two indices (n,l). Changes in shape for a given composition, changes the location of the "shape poles." Symmetric pole patterns with respect to the imaginary axis (Im x) are indicative of bodies close to the spherical shape. Asymmetric pole patterns with respect to Im x are indicative of elongated cylindrical shapes viewed at normal incidence. The peculiar pole asymmetry of the cylindrical shape is due to the presence of an additional branch of poles that shifts the symmetry of all the other poles. A study of these poles is presented indicating: (1) How to distinguish poles due to body shape from those due to composition; (2) How the pole positions change as the composition changes from the very stiff to the very compliant limits for a given shape; (3) How to look for the identifying symmetries or asymetries; (4) How to recognize the shape when only a limited number of poles are available; and (5) How noise levels make the pole-positions random, and known only within "uncertainty circles."

9:15

KK2. Backscattering of light from spherical and slightly spheroidal air bubbles in water: A novel unfolding of the glory. W. Patrick Arnott and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

The optical glory, which is related to a weak focusing of backscattered light, was observed for bubbles in water. This caustic is affected by small deviations from sphericity. Hydrodynamics causes freely rising bubbles in pure water to become slightly spheroidal if their diameter $\gtrsim 250\,\mu$m. We modeled and photographed the near backward scattering from spherical and slightly spheroidal horizontally illuminated bubbles. Spherical bubbles produce backward-directed glory wavefronts having the local shape of a circular torus [D. S. Langley and P. L. Marston, Phys. Rev. Lett. **47**, 913–916 (1981)]. Our model introduces the leading perturbation of those wavefronts resulting from a small deviation from sphericity. The cross-polarized scattering from a sphere has a fourfold symmetric azimuthal dependence; the patterns for spheroids are typically twofold symmetric. Detailed features of the modeled patterns are confirmed by the observations. This gives a specific example of broken symmetry resulting in an unfolding of a caustic which would otherwise have an infinite co-dimension. Related unfoldings should occur in certain acoustical scattering problems. [Work supported by ONR.]

9:30

KK3. Hyperbolic-umbilic diffraction catastrophes and the tracing of local principal curvatures of wavefronts. P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

Focal sections characteristic of a hyperbolic-umbilic diffraction catastrophe (hudc) are known to be observable in light scattered from spheroidal drops [P. L. Marston and E. H. Trinh, Nature **312**, 529–531 (1984)]. They should also be present in high-frequency sound reflected from curved surfaces or refracted by inhomogeneities. It is shown that a wave whose amplitude in the xy plane is $\exp[ik(g - ct)]$, will produce a far-field hudc focal section if $g(x,y) = \alpha x^3 + 3\gamma y^2 x)/6$, where α and γ are related to the local principal curvatures (k_1 and k_2) of g for $y = 0$ by $k_1 = \alpha x$ and $k_2 = \gamma x$. The apex angle Ψ formed by the caustics of the hudc is given by $\tan(\Psi/2) = (\gamma/\alpha)^{1/2}$. Therefore, to predict the hudc pattern k_1 and k_2 are needed for points on the outgoing wavefront adjacent to the focal ray at $x = 0, y = 0$. These may be found by tracing up to the xy plane the local principal curvatures of the wavefront. This procedure was verified for the aforementioned light scattering problem; in agreement with data on acoustically levitated drops, it gives $\tan(\Psi/2) = m/(12)^{1/2}$, where m is the drop's refractive index. [Work supported by ONR.]

9:45

KK4. Coupled finite element and boundary element capability for acoustic scattering from general structures. Gordon C. Everstine (David Taylor Naval Ship R&D Center, Bethesda, MD 20084) and Louise S. Schuetz (Naval Research Laboratory, Washington, DC 20375)

A coupled finite element/boundary element capability is described for calculating the sound pressure field scattered by an arbitrary submerged 3-D elastic structure. Structural and fluid impedances are calculated with no approximation other than discretization. The surface fluid pressures and velocities are first calculated by coupling a finite element model of the structure with a discretized form of the Helmholtz surface integral equation for the exterior fluid. Farfield pressures are then evaluated from the surface solution using the Helmholtz exterior integral equation. The formulation is validated using known analytic solutions for scattering from submerged spherical shells.

10:00

KK5. Aspect ratio dependence of forward scattered form functions. Guy Norton and M. F. Werby (NORDA, NSTL, MS 39529)

It is usual to examine backscattered form functions when scattering from submerged objects. For such calculations frequency is allowed to vary over a broad range in small step sizes yielding resonance information and circumferential diffraction effects. For ka values approximately 5 or greater (here k is the wavenumber and a is a characteristic dimension) the form function remains at a fixed value with pronounced fluctuations at resonance locations. Forward-scattered form functions, however, continue to increase in value. Though they are not apparently as useful in the extraction of resonance information, calculations presented here indicate that the rate at which they increase with increasing frequency may be associated with the aspect ratio of the target, providing one limits calculations to end-on incidence.

10:15

KK6. Numerical approach to acoustical scattering from objects in a waveguide. M. F. Werby and Guy Norton (NORDA, NSTL, MS 39529)

Scattering from submerged objects in a free environment has been a problem of considerable interest for the past several years. Several approaches have yielded interesting results due to continued improvements in numerical schemes. The problem of scattering in a confined environment has proven more difficult to formulate in a form useful for calculation due to the coupling of effects from the scattered object with that of the boundaries. The purpose of this work is to propose a numerical scheme that enables one to describe adequately scattering from an elongated object in an environment capable of producing a guided wave. The method is then applied to scattering from an elongated rigid spheroid in a waveguide.

10:30

KK7. Comparison of scattering from elastic shells and end-on incident scattering from elastic spheroidal shells. M. F. Werby (NORDA, NSTL, MS 39529) and L. H. Green (Martin Marietta Baltimore, Baltimore, MD 21220)

Until fairly recently it has not been possible to perform exact calculations of scattering from nonspherical three-dimensional elastic shells. However, due to the extended boundary condition method (EBC) of Waterman, such calculations are presently possible. Both as a check of current numerical codes, and because of an interesting physical observation, a comparison of scattering from submerged spherical shells, and end-on incident scattering from spheroidal shells of the same material are presented in this work. The spherical calculations are performed using a normal mode code and the spheroidal calculations are performed using a code based on the EBC method for aspect ratios ranging from 1 to 3 for two different shell thicknesses. A comparison of the spheroidal form function with that of the spherical case illustrates a one-to-one correspondence between the two in which the form function of the spheroidal cases appear as stretched spherical calculations. This stretching effect is easily explained using the standing wave theory of resonances proposed by Werby and Gaunaurd.

10:45

KK8. Acoustic backscatter from highly dense scattering arrays. Kenneth L. Telschow (Idaho National Engineering Laboratory, Idaho Falls, ID 83415)

Acoustic backscatter signals from several specimens containing high number densities of small spherical scatterers were measured in order to compare the signals obtained with those calculated from the independent scatterer model for the case of high scatterer concentration and long wavelengths. The scatterers were prepared by precipitation of less dense second-phase spherical particles from a glass host by controlling the melt chemistry and cooling rate. Statistics on the size, number, and distribution of the scatterers were obtained from microscopic observation. The results show that the backscatter signal energy measured is directly related to the mean interparticle spacing between scatterers. The reason for this relationship is shown, from the independent scatterer model, to come from the fact that the distribution of interparticle spacings can be represented by a Gaussian probability with a specified mean. Results of the calculation and comparison with experimental results illustrating the effect of the spacing distribution on the scattering structure factor are presented. [Work supported by the Interior Department's Bureau of Mines through Department of Energy.]

11:00

KK9. Backscattering of leaky Lamb waves from fiber-reinforced composite laminates. Alain Jungman,[a] Peter B. Nagy, and Laszlo Adler (Department of Welding Engineering, The Ohio State University, 190 West 19th Avenue, Columbus, OH 43210)

Fiber-reinforced composite laminates were studied by analyzing the backscattered amplitude spectra of an incident broadband pulse. An extensive spatial averaging was carried out in the frequency domain to identify the coherent backscattering signals. For various angles of incidence the received signal was spectrum analyzed by using a digitized system. A set of curves were generated that clearly identified the source of backscattering as backscattered leaky Lamb waves. Results were generated by this method for different orientations of the fibers and compared to other ultrasonic techniques. [a] Permanent address: Universite Paris 7, 2 place Jussieu, Paris Cedex 05, France.

11:15

KK10. Scattering of acoustic waves by piezoelectric composites. V. K. Varadan, Y. Ma, and V. V. Varadan (Laboratory for Electromagnetic and Acoustic Research, Department of Engineering Science and Mechanics and Center for Engineering of Electronic and Acoustic Materials, The Pennsylvania State University, University Park, PA 16802)

Piezoelectic composite materials can be designed so that they effectively alter the reflection and transmission characteristics of acoustic and electromagnetic waves that are incident on them. This can be achieved by controlling their microstructure so that their performance characteristics match desired goals. With this end in mind basic research has been performed on the propagation of elastic *P*, *SV*, and *SH* waves and electromagnetic waves propagating through a composite containing a random distribution of parallelly aligned, infinitely long cylinders made of a piezoelectric material that are embedded in a host material such as rubber. If elastic waves are incident on such a medium, both elastic and electromagnetic waves are produced due to the piezoelectric effect. We use a self-consistent multiple scattering theory that takes into account the interacting elastodynamic and electromagnetic fields as well as statistical correlations among the cylinders that enter the expressions for the average fields in this medium. The response of a single piezoelectric cylinder to applied elastic and electromagnetic fields is characterized by a T matrix. From the configurationally averaged fields, the effective elastic, electromagnetic, and piezoelectric properties can be obtained. In our computations, $BaTiO_3$ fibers embedded in a rubberlike host material with different rigidities are considered. Numerical results are presented for both coherent phase velocity and attenuation as a function of frequency and concentration of the piezoelectric cylinders. To evaluate the piezoelectric effect in altering the propagation characteristics of the waves, the numerical results obtained are compared to those obtained suppressing this effect.

11:30

KK11. Plane wave scattering from compliant tube gratings at oblique incidence. C. Audoly and P. Lamy (G.E.R.D.S.M., Le Brusc, 83 140 Six-Fours, France)

A computer model, based on a method presented by previous authors [R. P. Radlinski and M. M. Simon, J. Acoust. Soc. Am. **72**, 607–614 (1982)] and extended to oblique incidence has been developed to calculate scattering of a plane wave from compliant tube gratings in water. Insertion loss and total pressure field at both sides of the gratings are given and compared to experiments. The cases of single- and double-layer gratings embedded in a viscoelastic layer or not are considered. For some cases of incidence, the excitation of a class of resonant modes of the tubes section causes significant changes in insertion loss curves. When the gratings are encapsulated the effects of incidence are less important.

11:45

KK12. Numerical search for a breakdown of the Waterman algorithm for approximating the *T* matrix of a rigid obstacle. Angie Sarkissian (Sachs/Freeman Associates, Landover, MD 20785), Allan G. Dallas (Naval Research Laboratory, Code 5132, Washington, DC 20375-5000), and Ralph E. Kleinman (Department of Mathematical Sciences, University of Delaware, Newark, DE 19711)

The convergence of the Waterman algorithm for approximating the T matrix [J. Acoust. Soc. Am. **45**, 1417–1429 (1969)] has not been established for scatterers of general shape. One can prove that the convergence criteria of Kristensson, Ramm, and Ström [J. Math. Phys. **24**, 2619–2631 (1983)] are never fulfilled if the obstacle is nonspherical and the spherical wavefunctions have their usual normalization; the proof of this fact shall appear elsewhere. One of the coordinate sequences of this algorithm is the family of regular spherical waves, which fails to be complete in $L_2(\Gamma)$, Γ denoting the obstacle boundary, when k^2 is an interior Dirichlet eigenvalue for the negative Laplacian. These circumstances lead one to conjecture that the method will fail at such frequencies (as it does for the sphere). The results of computations near various of these frequencies for spheroidal scatterers are reported. For the sphere, the widths of the ka intervals of instability are examined, while for the nonspherical case, the object is to determine whether a breakdown occurs at all.

Session LL. Psychological and Physiological Acoustics V: Animal Psychophysics and Biochemical Processes

Peter Narins, Chairman
Department of Biology, UCLA, Los Angeles, California 90024

Contributed Papers

8:30

LL1. Comparison of simulated peripheral auditory responses to vowels in noise with neural data. Karen L. Payton [a] (Electrical Engineering and Computer Science Department, Johns Hopkins University, Baltimore, MD 21218)

A model of the auditory periphery, designed to fit neural responses to tones, has been shown [K. L. Payton, J. Acoust. Soc. Am. Suppl. 1 77, S80 (1985)] to predict many response characteristics of auditory-nerve fibers to vowel sounds in quiet [M. B. Sachs and E. D. Young, J. Acoust. Soc. Am. 68, 858–875 (1980)]. This paper examines the effects of background noise on model vowel responses and compares them to neural responses [M. B. Sachs, H. F. Voigt, and E. D. Young, J. Neurophysiol. 50, 27–45 (1983)]. In quiet, both the neural and model average rate responses saturate at high stimulus levels. Representation of vowel formants by the ALSR, a temporal measure of neural responses relatively insensitive to level, is also predicted by the model. When white noise is added, at + 10 dB S/N ratio, both the neural and model average rate responses saturate at lower vowel sound levels. However, the presence of background noise degrades the model ALSR responses more than it degrades the neural ALSR responses. [Work supported by NSF.] [a] Present address: R.L.E. 36–747, MIT, Cambridge, MA 02139.

8:45

LL2. Vowel discrimination in primates. Joan M. Sinnott and Nancy A. Kreiter (Department of Psychology, Indiana University, Bloomington, IN 47405)

Discrimination of ten steady-state synthetic English vowels was measured in Old World monkeys (*Macaca, Cercopithecus*) and humans using a repeating standard AX procedure and positive-reinforcement operant conditioning techniques. All monkeys had difficulty discriminating spectrally similar vowels such as |ɑ-ʌ| and |ν-μ|, but macaque monkeys were superior to cercopithecus monkeys. Human reaction times reflected spectral similarity and correlated with monkey percent correct discriminations. Macaque DLs for formant frequency changes along vowel continuua |ɛ-æ| and |i-ɪ| were about 30 Hz, compared to about 10 Hz for humans. Formant filtering and sensation level variation had only minor effects on DLs. Results are related to other comparative psychoacoustic data and vocal communication in the various monkey species. [Work supported by NIH and the Deafness Research Foundation.]

9:00

LL3. Auditory sensitivity of two strains of the common canary (*Serinus canarius*). Robert J. Dooling and Kazuo Okanoya (Psychology Department, University of Maryland, College Park, MD 20742)

Adult canaries from two strains of canaries (Belgian Waterslager and German Roller) were trained with operant conditioning techniques for audiometric testing. Absolute thresholds for canaries from the German Roller strain were similar to those previously reported for other small birds including canaries. At frequencies above 2.0 kHz, absolute thresholds for canaries from the Belgian Waterslager strain were 30–50 dB higher than those of German Roller canaries. The spectral distribution of energy in the calls of the two species parallels the species differences in absolute thresholds. Absolute thresholds of $F1$ hybrids of a male Belgian Waterslager and a female German Roller canary also show elevated high-frequency thresholds. These findings suggest that strain differences in auditory thresholds in the canary may have a genetic basis. The results are discussed in relation to selective breeding and song learning in domestic canaries. [Work supported by NIH.]

9:15

LL4. The relationship between sexual and individual differences in lovebird calls. Leslie A. Wheeler (Laboratoire de Physiologie Acoustique, INRA-CNRZ, F78350 Jouy-en-Josas, France and Naval Ocean Systems Center, Kailua, HI 96744), Jacques Lefebvre, and Michel Wimitzky (Laboratoire de Génétique Factorielle, INRA-CNRZ, 78350 Jouy-en-Josas, France)

Those of us who have tried to determine how acoustic information is transmitted in parrot calls have come up against a curious problem: while we have a pretty good idea of how parrots can use their calls to recognize each other as individuals, we cannot figure out how they recognize each other sexually. To try to solve this problem, our team changed strategy: if direct approaches fail, try an indirect approach. In short, to study sexual differentiation, we worked backwards and sideways from studies of individual differentiation. By putting together a jig saw puzzle whose pieces represent the results of psychoacoustic experimentation, of classical acoustic analysis, and of original techniques for acoustic information extraction, an image did indeed emerge and is presented in this paper.

9:30

LL5. Anatomical effects of intense tone stimulation in the ear of bony fishes. Mardi Cox, Peter H. Rogers (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332), Arthur N. Popper, and William M. Saidel (Department of Anatomy, School of Medicine-School of Dentistry, Georgetown University, Washington DC 20007)

In an attempt to determine whether or not a place-type mechanism for frequency discrimination exists in the ears of bony fishes, an experimental investigation of three species was conducted. Goldfish, Oscars, and Kissing Gouramis were exposed to intense single-frequency sound fields for 2 h and then sacrificed. During the exposure, the fish were constrained inside a waveguide at a point of maximal acoustic pressure and minimal particle velocity. The saccular maculae were examined under a scanning electron microscope to determine the location of hair cell damage as a function of frequency. Preliminary results indicate that the location of the damage may not depend on frequency. [Work supported in part by ONR and NIH.]

9:45

LL6. Auditory duration discrimination in primates. Joan M. Sinnott and Michael J. Owren (Department of Psychology, Indiana University, Bloomington, IN 47405)

Auditory duration DLs at 2 kHz were measured in Old World monkeys (*Macaca, Cercopithecus*) and humans using a repeating standard AX procedure and positive reinforcement operant conditioning tech-

niques. For a 200-ms standard tone, monkey DLs (45–62 ms) were about 2–3 times larger than human (15–27 ms), Weber fractions ($\Delta T/T$) for all species were smallest at standard durations of 200 and 400 ms and increased with decreasing standard duration. DLs for all species did not vary as a function of sensation level from 30 to 60 dB, but were elevated slightly at 20 dB. Monkeys experienced difficulty in discriminating duration decrements, in contrast to humans. Results are related to other comparative psychoacoustic data and vocal communication. [Work supported by NSF and the Guggenheim Foundation.]

10:00

LL7. Temporal limits of frequency discrimination in the components of two Bobwhite Quail (*colinus virginianus*) calls. R. W. Gatehouse (Department of Psychology, University of Guelph, Guelph, Ontario N1G 2W1, Canada)

Previous studies using operant training/testing regimes (Barton *et al.*, 1984; Bailey *et al.*, 1985) have determined audibility curves and frequency discrimination capabilities of the Bobwhite Quail (*colinus virginianus*) with a view to delineating the functional significance of various components in the birds' calls. The present studies required two groups of four birds to discriminate between predominant frequencies (1.0 and 3.5 kHz) which are significant in the adult separation and chick "lost" calls, respectively, under varied peak frequency temporal durations. Three studies were done: (1) the rewarded "group correct" signals (e.g., 1.0 kHz) and the competing signals (e.g., 3.5 kHz) durations were successively and equally decreased; (2) the competing signal was held constant (200 ms) while the group correct was decreased; (3) each "correct" frequency of 200 ms was compared against itself with durations less, equal, or greater than 200 ms. The results suggest that in the field where the quail must extract species specific information about conspecifics from the "noise" of other similar/dissimilar signaling (cross species and environmental noises) and from the similar frequencies, durations, and sequences put out by within species competitors, the signals must be at least 100 ms to have salience. The significance of such call component analysis in bird communciaton and survival will be discussed.

10:15

LL8. Frequency and intensity dependence of the preferred firing phase of starling cochlear ganglion cells. Peter M. Narins (Department of Biology, University of California, Los Angeles, CA 90024) and Otto Gleich (Institut für Zoologie, T. U. München, Lichtenbergstr. 4, 8046 Garching, Federal Republic of Germany)

The preferred phase of firing of 28 cochlear ganglion cells in the anesthetized starling, was measured over a frequency range encompassing CF of typically 3 octaves and an intensity range of at least 30 dB. The phase versus frequency functions could be fit remarkably well by straight lines; the slopes of these functions represent the system delay between the stimulus onset and the occurrence of the spike at the recording electrode. After correction for acoustic and neural transmission, we find delays for tone stimuli of 90 dB SPL between 1.37 ms (for the highest-CF cells) and 2.40 ms (for the lowest CF cells). In addition, the slopes of the phase versus frequency functions, and hence the system delay, decrease with increasing stimulus intensity by 0.04 ms/dB. As in amphibians, reptiles, and mammals, the preferred phase of firing is nearly intensity-independent near CF, whereas increasing stimulus intensity above and below CF results in a progressive phase lead and lag, respectively. These results are discussed with respect to the underlying mechanisms of frequency analysis in the avian inner ear. [Work supported by the A.v. Humboldt Foundation, the SFB, and NIH.]

10:30

LL9. Static and dynamic differences in perilymphatic potassium between scala tympani (ST) and scala vestibuli (SV). Kenji Ohyama, Eiichi Arakawa, Alec N. Salt, and Ruediger Thalmann (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

Cochlear perilymph has been regarded as a single, homogeneous fluid, although K is usually found in lower concentration in ST (3–4 mM) than SV (6–7 mM). Since perforation of the cochlea (to aspirate a sample or insert an ion selective electrode) releases cochlear pressure, allowing cerebrospinal fluid (CSF) (3 mM K) to enter, it has not been established whether the K difference is genuine or artifactual by contamination with CSF. In guinea pigs (GP) stringent precautions were used to prevent perilymph leakage and release of cochlear pressure during insertion of the ion selective electrodes. We found that K in ST is significantly lower than in SV, demonstrating for the first time a systematic *physiological* difference between the scalae. We have also found marked differences between the clearance of K in ST and SV following high K perfusion (20 mM) of guinea pigs (GP) with the cochlear aqueduct occluded. In general, the rate of K clearance from SV is 2–3 times slower than the clearance from ST. These data demonstrate that dynamic processes of perilymph homeostasis are not uniform throughout the cochlea and that concentration differences can be maintained across the spiral ligament. [Work supported by NIH and NSF.]

10:45

LL10. Marked differences in blood/perilymph kinetics of furosemide (FU) between scala tympani (ST), and scala vestibuli (SV). Paradoxical effect of probenecid (PR). Akira Hara, Ruediger Thalmann, and Alec N. Salt (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

Rybak *et al.* [J. Pharmacol. Exp. Therapeut. **230**, 706–709 (1984)] demonstrated that i.v. injection of 100 mg/kg FU results in a rapid but transient accumulation of the drug in ST of the chinchilla (CH). We found a similar pattern in ST of guinea pigs (GP) (5 μg/ml, 15 min after 100 mg/kg i.v.), followed by a (gradual) decline to 0.8 mg/ml within 1 h. In SV (not studied by Rybak *et al.*) FU increased very slowly (0.8 μg/ml at 15 min) but continued to rise gradually (2 μg/ml at 1 h). Rybak *et al.* found that 50 mg/kg PR suppressed accumulation of FU, while we found only much higher dosages (200 mg/kg) to be effective in GP. Paradoxically, at low dosages (50 mg/kg) PR caused an increase of FU that persisted for prolonged periods. This suggests that the influx and outflux systems for FU at the blood/perilymph barrier of ST have a differential sensitivity to PR, perhaps analogous to the differential modification of methotrexate flux in and out of tumor cells in response to PR [Henderson and Zevely, J. Biochem. Pharmacol. **34**, 1725–1729 (1985).]

11:00

LL11. Biochemical and biophysical properties of the tectorial membrane (TM). Isolde Thalmann, Gertraud Thallinger, and Ruediger Thalmann (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

It was recently reported [I. Thalmann *et al.*, ORL 48, 107–115 (1986)] that 40%–50% of the guinea pig TM consists of collagen. We now provide further rigorous proof that most of the collagen consists of type II, and that also the minor cartilage collagens, 1α, 2α, and 3α, as well as collagen type IX are present. Thus, the TM contains the same types of collagen as hyaline cartilage. Moreover, we have shown by quick-freeze, deep-etch 3-D electronmicroscopy (3-D-EM) that at least the upper part of the TM consists of a dense tangle of fibrils ($\sim$20 nm) which are indistinguishable from the type II collagen fibrils of cartilage. These fibrils are embedded in a dense meshwork of anastomotic material that looks similar to the proteoglycan gel observed throughout cartilage but which differs from cartilage in that its constituent strands look relatively thick and variably puffy. Although the majority of the noncollagenous proteins are also glycoproteins, it cannot yet be decided whether and to what extent they contribute to a second type of major fibril or to the gel matrix. The presentation will include a description of the dry weight of the TM as a function of cochlear location and an estimate of the dry-to-wet weight ratio. [Work supported by NSF; We thank Dr. J. Heuser for the 3-D-EM preparations.]

11:15

LL12. Some biochemical and physicochemical features of the spiral limbus–basilar membrane–spiral ligament complex. Ruediger Thalmann, Heinz J. Kellner, Thomas H. Comegys, and Isolde Thalmann (Department of Otolaryngology, Washington University, Saint Louis, MO 63110)

It has been shown by immunohistochemical techniques that the predominant collagen of inner ear connective tissues is collagen type II [K. Tomoda *et al.* Ear Res. Jpn. **15**, 199–202 (1984)]. It has now been confirmed by chemical techniques that collagen type II is also abundant in spiral limbus (LIM), basilar membrane (BM), spiral ligament (LIG); in addition, the "minor cartilage collagens" (1α, 2α, 3α) and collagen type IX are present. No collagen was found in the stria vascularis (SV). In contrast to the situation with LIM and LIG, it was extremely difficult to extract collagen from BM, presumably because the fibers are embedded in a thick, dense "ground substance." BM was homogenized vigorously with ultrasound and glass micro-beads, then digest with pepsin in order to release the collagen. Protein content of BM ranged from 21% to 28% of dry weight, in rough correspondence with the well-known differences in structural detail of the BM along the extent of the cochlea. In turn, collagen comprises from 36% to 44% of the total protein content. BM also contains appreciable levels of OCP-I and OCP-II, polypeptides previously assumed to be organ of Corti-specific. OCP-I and II are also present in very low concentrations in LIM and LIG, but not in SV. The significance of these findings will be discussed. [Work supported by NSF.]

THURSDAY MORNING, 11 DECEMBER 1986 — SALON E, 8:30 A.M. TO 12:30 P.M.

Session MM. Speech Communication V: Speech Perception by the Hearing Impaired; Pathological Voice

Robert J. Porter, Jr., Chairman
Kresge Hearing Research Laboratory of the South, Department of Otorhinolaryngology and Biocommunication, LSU Medical School, Florida Avenue, New Orleans, Louisiana 70112

Contributed Papers

8:30

MM1. Identification of vowels in noise and in reverberation produced by a female talker. Anna K. Nábělek (Department of Audiology and Speech Pathology, The University of Tennessee, Knoxville, TN 37996-0740)

Vowels produced by a female talker were less identifiable than previously investigated vowels produced by a male talker [A. K. Nábělek and P. A. Dagenais, J. Acoust. Soc. Am. **80**, 741–748 (1986)]. Matrices of responses were obtained for normal-hearing and hearing-impaired subjects for three conditions: quiet, noise (S/N = 0 dB), and reverberation ($T = 1.2$ s). The vowels which were confused by normal-hearing subjects were also confused by hearing-impaired subjects. However, there were some pairs of vowels which were confused only by hearing-impaired subjects. There were also some differences between errors for the female voice and errors collected previously for the male voice. Differences in errors for the two talkers and two groups of subjects will be discussed. [Work supported by NIH.]

8:42

MM2. Effects of aging and hearing loss on sentence reception in noise from one versus two sources. Stanley A. Gelfand, Leslie Ross (Audiology and Speech Pathology Service (126), Veterans Administration Medical Center, East Orange, NJ 07019), and Sarah Miller (City University of New York, New York, NY 10036)

Duquesnoy [J. Acoust. Soc. Am. **74**, 739–743 (1983)] found that elderly subjects with hearing loss do not take advantage of the separation of speech and noise sources to the same degree as young normals. However, it is not clear whether this finding was due to aging and/or the hearing loss. We addressed this issue using a sentence reception threshold (SRT) test developed by modifying the SPIN test, which produces data comparable to those obtained with the Dutch sentences. Here SRTs were obtained from young and elderly normal hearing subjects and an elderly hearing impaired group using sentences and noise presented from one speaker or from two speakers 90° apart. The impaired group performed more poorly than the normal groups in all conditions. The advantage derived from separating the speech and noise compared to presenting them from the same speaker was smaller for the impaired group than for the normal groups. The normal groups did not differ significantly. The reduced ability of the elderly hearing impaired subjects to take advantage of the separation of speech and noise appears to depend more on the hearing loss than aging, per se. [Work supported by the Veterans Administration.]

8:54

MM3. Vowel identification and vowel masking patterns of hearing-impaired subjects. Dianne J. Van Tasell, David A. Fabry (Department of Communication Disorders, 115 Shevlin Hall, University of Minnesota, MN 55455), and Linda M. Thibodeau (Department of Speech Communication, University of Texas, Austin, TX 78712)

Confusion matrices for seven synthetic steady-state vowels were obtained from ten normal and three hearing-impaired subjects. The vowels were identified at greater than 96% accuracy by the normals, and less accurately by the impaired subjects. Shortened versions of selected vowels then were used as maskers; vowel masking patterns (VMPs) consisting of forward-masked thresholds for sinusoidal probes at all vowel masker harmonics were obtained from the impaired subjects and from one normal subject. Here VMPs of the impaired subjects, relative to those of the normal, were characterized by smaller dynamic range, poorer peak resolution, and poorer preservation of the vowel formant structure. These VMP characteristics, however, did not necessarily coincide with inaccurate vowel recognition. Vowel identification appeared to be related primarily to VMP peak frequencies rather than to relative peak amplitudes or between-peak characteristics of the patterns. [Work supported by NINCDS.]

9:06

MM4. Filtering competing messages to enhance mutual intelligibility. C. R. Corbett, P. M. Zurek, N. I. Durlach, and W. M. Rabinowitz

(Research Laboratory of Electronics, MIT, Room 36-730, Cambridge, MA 02139)

Filtering simultaneous messages into different frequency bands prior to summation and presentation to a single ear was examined as an aid to message reception. Two or four concurrent sentences spoken by different talkers were presented through a single earphone to normal-hearing subjects. Listeners were instructed to write down the message spoken by the target talker, whose voice was indicated by an isolated lead-in message. In the reference condition, the wideband (0–4.5 kHz) sentences were simply summed; in various test conditions, the sentences were passed separately through different filters and then summed. Average intelligibility of the filtered sentences in the two-message task did not exceed that of the reference condition for any scheme tested. In the four-message task, however, performance improved significantly for three of five filtering configurations evaluated, the best of which resulted in 39% correct overall intelligibility compared to approximately 22% correct in the reference condition. Such a filtering scheme may be beneficial in future hearing aids that present multiple directional channels to a single ear. [Work supported by NIH.]

9:18

MM5. The perception of temporally modified conversational and clear speech by hearing-impaired listeners. R. M. Uchanski, L. D. Braida, N. I. Durlach, and C. M. Reed (Research Laboratory of Electronics, MIT, Room 36-749, Cambridge, MA 02139)

Hearing-impaired listeners find clearly articulated speech more intelligible than conversational speech [M. A. Picheny *et al.*, J. Speech Hear. Res. **28**, (1985)]. Acoustical measurements indicate that the temporal characteristics of clear and conversational speech differ [M. A. Picheny *et al.*, J. Speech Hear. Res. (in press)]. In particular, clearly articulated segments are nonuniformly longer than the corresponding segments in conversational speech [R. M. Uchanski *et al.*, J. Acoust. Soc. Am. Suppl. 1 **77**, S54 (1985)]. For example, the percent increase in duration for fricatives and diphthongs is much larger than that for short vowels and voiced plosives. To evaluate the importance of these temporal differences we have measured the intelligibility of two types of processed speech materials: conversational speech with nonuniformly increased segment durations and clear speech with nonuniformly compressed segment durations. Results from several hearing-impaired listeners will be discussed. [Work supported by NIH.]

9:30

MM6. Spectral compression, a processing scheme for single-channel sensory aids. Richard R. Hurtig (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Single-channel vibro-tactile discrimination and identification of vowels was assessed using a scheme which maintains the spectral shape of the complex speech waveform. A series of vowels and diphthongs were synthesized using the Klatt algorithm. Unlike conventional synthesis, the formant frequencies were set such that the first four formants fell under 660 Hz and the formant bandwidths were narrowed. This synthesis effectively generated 5:1 frequency compression. The synthesized segments sound speech like. Naive subjects were presented both discrimination and identification tasks. The stimuli were felt with a Audiological Engineering V1220 transducer. With no prior exposure discrimination exceeded 7%, and within a few hours of testing discrimination levels of 95% were achieved. Subjects were able to accurately identify many of the vowel segments. Furthermore examination of the confusions reveals patterns similar to those reported for auditory confusions of natural vowels. These findings appear to contradict the premise underlying the development of multichannel tactile aids and cochlear implants that the tactile senses and the impaired ear are incapable of extracting the appropriate information from a single complex speech waveform.

9:42

MM7. Judgments of intonation and contrastive stress during lipreading. Lynne E. Bernstein, Silvio P. Eberhardt (Sensory Aids Laboratory, Department of Electrical Engineering, Johns Hopkins University, Baltimore, MD 21218), and Marilyn E. Demorest (Department of Psychology, University of Maryland Baltimore County, Catonsville, MD 21228)

As part of work to develop vibrotactile devices to convey voice fundamental frequency to hearing-impaired lipreaders, an experiment was conducted to investigate the visibility of contrastive stress and question-versus-statement intonation contours. Four sentences ("We will weigh you," "We owe you a yoyo," "Chuck caught two cats," and "Pat cooked Pete's breakfast") with contrastive stress on one of the first three words and spoken as either statements or questions by a male and a female were presented from videodisk. Sentences were chosen to minimize indexical or affective information that might be used to judge stress or intonation. Subjects were tested in a six alternative forced-choice procedure with response alternatives labeled "question" or "statement" and stress position "1," "2," or "3." Results suggest that contrastive stress and intonation can be judged visually at levels significantly above chance. Judgments of stress were significantly more accurate than judgments of intonation. Results will be discussed in terms of loglinear models to assess the relative independence of stress versus intonation in visual judgments. [Work supported by NIH.]

9:54

MM8. Transformations of voice fundamental frequency for a vibrotactile device to aid lipreading. Silvio P. Eberhardt and Lynne E. Bernstein (Sensory Aids Laboratory, Department of Electrical Engineering, Johns Hopkins University, Baltimore, MD 21218)

As part of the development of a wearable vibrotactile device to convey voice fundamental frequency F_0 to hearing-impaired lipreaders, a study was conducted to compare transforms of F_0. Rothenberg and Moliter [J. Acoust. Soc. Am. **66**, 1029–1038 (1979)] reported that most errors in a tactile-alone task of judging stress and intonation in sentences were mislocations of the stressed word. We found that contrastive stress can be detected visually. The present experiment compared F_0 transforms in a tactual–visual task. Transforms included: (1) direct glottal to tactile pulse; (2) linear transform with lowering of center frequency and shift in range; (3) similar to (2) with log of the normalized glottal period; (4) similar to (2) with the difference of logs of the current period and a weighted sum of previous periods; and (5) a constant pulse train during voicing. Subjects received tactile stimulation to the index finger by an AV-6 vibrator, and responses were labeled "question" or "statement" *and* one of three stress positions. Effects of the transforms and conditions of vision-only versus tactual–visual presentation will be reported. [Work supported by NIH.]

10:06

MM9. Effects of filter configuration on categorical perception of tactually presented speech. Rebecca E. Eilers, D. Kimbrough Oller, Edward Miskiel, Debra Moroff (Department of Pediatrics, University of Miami, P. O. Box 016820, Miami, FL 33101), and Ozcan Ozdamar (Department of Biomedical Engineering, University of Miami, P. O. Box 016820, Miami, FL 33101)

The effect of filter configuration on the relationship between auditory and tactual perception of speech was investigated via a 32-channel computer controlled electrocutaneous display and normal audition. Two 11-step synthetic speech continua, /a/ to /ə/ and /sta/ to /sa/, served as stimuli for study of three filter configurations: logarithmic, linear, and average (arithmetic mean of log and linear). Four well-practiced subjects performed two tasks with each stimulus continuum and each filter configuration: (1) a standard identification procedure where all 11 steps are categorized as either endpoint 1 or 11, and (2) a pairwise discrimination task (where equal interval stimuli were discriminated across the continuum). The same tasks were presented in the auditory modality for comparison purposes. Results indicate (1) a close correspondence between tactual and auditory perception, (2) categorical perception of the consonontal continuum and more continuous perception of the vowel continuum in both modalities, and (3) an interaction between filter configuration and speech syllable type that affects the degree of similarity between auditory and tactual functions. Implications in terms of theories of speech perception will be discussed. [Work supported by NIH.]

10:18

MM10. A sentence test for minimizing learning effects. Harry Levitt and Arlene C. Neuman (Graduate School, City University of New York, 33 West 42 Street, Room 902, New York, NY 10036)

Speech tests are often used for the evaluation of hearing aids. When several hearing aids are being compared, it is necessary to have lists of equivalent difficulty or lists which can be used repeatedly. Although sentences provide a much closer approximation to real-life communication than single word or syllable tests, sentence materials are seldom used for repeated testing because of the substantial learning effects involved. Typically, a sentence list is used only once with each subject. Testing with a closed response set reduces learning effects considerably and a test of this type has been developed for sentences using a true–false response format. Each list consists of balanced sets of sentences in which the meaning of each sentence has been altered by a combination of two or three simple changes; e.g., subject/object reversal, use of negation, active/passive transformation. Repeated testing with the same sentence list (order of test items randomized) showed small to negligible learning effects. The test also provided information on relative performance with different syntactic forms. [Research supported by NINCDS.]

10:30–10:42

Break

10:42

MM11. Striking differences in the babbling of deaf and hearing infants. D. K. Oller and Rebecca E. Eilers (Department of Pediatrics, University of Miami, Mailman Center for Child Development, P. O. Box 016820, Miami, FL 33101)

It is widely believed that deaf infants "babble" in much the same way hearing infants do, although the deaf have been said to produce less babbling than the hearing after six or seven months of age. These beliefs have been based upon sketchy and sparse data. Over the past 14 years, we have been collecting tape-recording samples of vocalizations and now have extensive recordings from 8 (severely and profoundly) deaf infants who are otherwise unimpaired and 21 hearing infants. The hearing babies all began canonical babbling (production of well-formed syllable sequences such as [bababa], [nanana], etc.) by 10 months of age while not one of the deaf babies started canonical babbling before 11 months. The onset of canonical babbling is considered significant because it represents the point at which infants manifest a capacity to produce syllables that meet all the requirements of linguistic–phonetic units. The difference in onset of babbling in the two groups of infants is reliably identified by adult listeners.

10:54

MM12. Jaw kinematics in hearing-impaired speakers. Nancy S. McGarr, Anders Lofqvist, and Robin Seider Story (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

This study examines jaw movements as a function of vowel height and stress in real-word phrases produced by three hearing-impaired speakers and a hearing control. Movements of the lips and jaw were recorded using an optical tracking system. Measures included duration, displacement, and peak velocity of movement. There were statistically significant main effects for vowel height in nearly all movement records for both the hearing-impaired and the hearing speakers. However, the related measures for stress had no significant effects for the subjects. The hearing speaker distinguished stressed and unstressed segments by maintaining the jaw in a lowered position for a longer period in the stressed vowels. With few exceptions, kinematic values for the hearing and the hearing-impaired speakers were comparable. These results thus suggest that some characteristics of speech production are not adversely affected by congenital and profound hearing loss. [Work supported by NIH.]

11:06

MM13. Using visual targets to model and shape /s, ɾ/ sound production by a deaf talker. Samuel G. Fletcher (Department of Biocommunication, University of Alabama at Birmingham, Birmingham, AL 35294)

This paper will report on a training program that enabled a 17 year old, profoundly hearing-impaired girl to produce /s/ and /ɾ/ successfully for the first time. Auditory and electropalatometric procedures were used to verify the absence of recognizable sibilant sounds in her pretreatment articulatory repertoire. The targets were an 8-mm central groove width for /s/ and a 12-mm more posteriorly located groove for the /ɾ/. The targets were displayed on a CRT array of 96 points in a grid pattern. The subject produced the sounds in three conditions: (1) without visual feedback, (2) with points highlighted as associated sensors were contacted, and (3) with overlaid highlighting of targeted points and intensification of highlighted points when the sensors were actually contacted. When nontargeted sensors were contacted, the talker was notified by less visible overlayed asterisks. At the end of each trial, the extent of successfully matching the targeted pattern was automatically calculated and numerically displayed. The degree of match achieved in isolated sounds—and later in words and sentences loaded with /s, ɾ/ sounds—increased steadily across the eight training sessions. Rapid learning of the articulatory patterns and emergence of auditorally and palatometrically identifiable /s/ and /ɾ/ sounds were found.

11:18

MM14. An integrated voice analyzer for evaluating pathological voice. Yoshinobu Kikuchi, Satoshi Uchida, and Hideki Kasuya (Department of Electronic Engineering, Faculty of Engineering, Utsunomiya University, Utsunomiya 321, Japan)

We have developed an integrated voice analyzer (IVA) using DSP (TMS32010) to achieve high-speed acoustic analyses of a sustained vowel. A personal computer connected to the IVA is employed as the man–machine interface as well as to display and save various analysis results. The IVA has many functions, such as detecting vocal noise by an adaptive comb filter, measuring period and amplitude perturbation quotients, calculating DFT spectra and LPC coefficients, etc. The results are stored into a disk file to do statistical analysis or to perform further analyses. This system provides an interactive acoustic analysis method for quantitative evaluations of the pathological voice arising in various voice research and clinical areas. We describe an application of the IVA for the clinical screening system for pathological voices, where hoarseness grade (HG) of a voice can be obtained from five sustained phonations. [Work supported by a Japanese Grant-in-Aid for Scientific Research.]

11:30

MM15. Perceptual significance of a vocal noise measure for the evaluation of pathological voice. Hideki Kasuya, Yoshinobu Kikuchi, and Kiyomasa Hasegawa (Department of Electronic Engineering, Faculty of Engineering, Utsunomiya University, Utsunomiya 321, Japan)

By using a comb filter, harmonic components in a sustained vowel have been separated from the vocal noise which is primarily associated with the turbulent noise produced at the glottis due to insufficient closure of the vocal folds [H. Kasuya *et al.*, Proc. ICASSP-86, 669–671 (1986)]. An acoustic measure, normalized noise energy (NNE) is then defined on the extracted noise signal, by measuring the relative level of the noise energy to that of the original voice. The NNE has shown better detectabil-

THURSDAY AM

ity of pathological voice status than jitter and shimmer parameters. This paper investigates the relation between the NNE and perceived qualities of hoarseness of a pathological voice, as "hoarseness grade" and "breathiness." From the experiments we conclude that the NNE is related to the perceptual qualities of a pathological voice much better than jitter and shimmer parameters and some other noise measures do. [Work supported by a Japanese Grant-in-Aid for Scientific Research.]

11:42

MM16. Pitch extraction of esophageal speech. Heung-Kuk Kim and Donald S. Cooper (Department of Otolaryngology and Speech Science and Technology Program, University of Southern California, Los Angeles, CA 90033)

A number of studies have been devoted to the description of the prosodic characteristics of esophageal speakers. However, pitch extraction of esophageal speech is difficult because of its low pitch, of the often quasiperiodic character of the vibration of the pharyngo-esophageal segment, and because stoma noise may be added to the signal. Commercially available hardware F_0 extractors often produce artefacts in the analysis of esophageal speech. Consequently it may be asked whether the harmonic source model of laryngeal speech is inappropriate for esophageal speech, or whether it simply constitutes an extreme of this model in terms of the proportion of noise it contains. In a number of recent studies, cepstral analysis has been applied to pitch extraction of esophageal speech. This procedure handles low fundamentals well, but is computationally slow and complex. In this study, several other pitch extraction algorithms which are computationally simpler than the cepstrum will be compared to cepstral analysis in regard to their adequacy for F_0 extraction with esophageal speech. Some of these, especially the harmonic product spectrum and the harmonic sum spectrum, have been shown to be superior to the cepstrum in the analysis of noise speech signals. [Work supported by the Norris Cancer Hospital, USC.]

11:54

MM17. Co-articulation in aphasic speech. Betty Tuller (Department of Psychology, Florida Atlantic University, Boca Raton, FL 33431, and Haskins Laboratories, New Haven, CT 06511), and Robin Seider Story (Communication Sciences, University of Connecticut, Storrs, CT and Haskins Laboratories, New Haven, CT 06511)

Co-articulation, the interleaving of articulatory movements for neighboring speech sounds, is ubiquitous in normal speech and its' perceptual advantages are well documented. Yet surprisingly little is known concerning the preservation or disruption of co-articulation in aphasic speech. In this study, five fluent and five nonfluent asphasics, and matched control speakers, produced two-word utterances of the form $CV_1(z)\#sV_2C$. Vowels included /i/ and /u/. Half the test sequences included a word-final /z/ (e.g., these suits), and half did not (e.g., pea soup). Multiple tokens of the eight word pairs were elicited. Analyses included only those tokens that listeners correctly identified. Here LPC spectra were computed for the fricative noise in each vowel context and spectral peaks associated with the vowel's second formant were measured at the fricative's midpoint. Control subject's spectral peaks varied with V1 and V2, as expected, and fluent aphasics showed this same pattern. In contrast, only one nonfluent speaker showed evidence of anticipatory or carryover coarticulation. For the rest, spectral peaks did not covary with vowel identity or with intervocalic duration. [Work supported by NINCDS grants to Haskins Laboratories and Cornell University Medical Center, and by a grant from the Stuttering Center, Baylor College of Medicine.]

12:06

MM18. Durational characteristics of the speech of normal elderly adults. Jan Wasowicz, Bruce L. Smith, and Judy Preston (Department of Communication Sciences and Disorders, Northwestern University, Evanston, IL 60201)

A number of physical and psychological changes occur as a result of normal aging. These changes often result in an increase in the time subjects require to perform various sensorimotor tasks. Although the effects of aging upon a variety of behaviors have been well documented, relatively little is known about how normal aging may affect speech motor control. The present study examined temporal characteristics of the speech of ten normal, elderly adults and ten young adults, who produced a variety of words and sentences at normal and fast speaking rates. Acoustical analyses indicated that the elderly adults' segment, syllable, and sentence durations were 20%–25% longer than those of the young adults at both speaking rates. It was also found that the durations of the older subjects' speech at the fast rate were approximately equal to those of the younger adults' speech at a normal rate. In addition to comparisons made between these two groups of subjects, comparisons were also made with durations of the speech of young children studied in previous research.

12:18

MM19. The acoustical signature for intelligibility test words: Population profiles for neurologically intact, geriatric speakers. Gary Weismer, Ray D. Kent, and Megan Hodge (Department Communicative Disorders, University of Wisconsin, Madison, WI 53706)

A common approach to the clinical assessment of disordered speech is to obtain intelligibility scores for lists of single-word or sentence items. Whereas such scores provide an index of speech deficit severity, they give no insight to the acoustic basis of the deficit. A test that would allow specification of the acoustical basis of an intelligibility deficit should (1) incorporate word items that sample systematically the acoustical contrasts that cue differences in meaning, and (2) have as a foundation an empirically based model of the "normal" acoustical characteristics for each word in the test. In the present paper, speech profiles of selected intelligibility words from a closed-set test are presented for a relatively large group of geriatric speakers ($N = 15$, 15 females). Consideration will be given to ways of representing the normal population space for various acoustical events of speech, and selected data from dysarthric speakers will be presented to illustrate the application of this approach to a clinical population. [Work supported by NIH.]

Session NN. Underwater Acoustics VII: Remote Sensing and Underwater Acoustics, Part 1

William A. Kuperman, Chairman
Naval Research Laboratory, Washington, DC 20375

Invited Papers

8:30

NN1. The application of present and future satellite remote sensors to oceanographic acoustics. R. A. Shuchman (Environmental Research Institute of Michigan, Ann Arbor, MI 48107)

An ensemble of active and passive satellite remote sensors are presently operating or will be launched in the near future that provide high resolution detailed oceanographic information on a synoptic scale. These sensors operate in the visible, infrared, and microwave region of the electromagnetic spectrum and include the NOAA 7/8 (TIROS), NIMBUS-7, LANDSAT, SPOT, and GEOS satellites presently in orbit as well as the planned ESA ERS-1, NROSS, TOPEX, RADARSAT, and JERS-1 satellite launches. The information provided by these satellites includes not only ocean surface and air/sea environmental information such as wind speed and direction, gravity wave spectral estimates, surface water height, salinity, temperature, and water vapor content, but also information about the interior of the water column (i.e., fronts, upwelling, and internal waves). Additional information about the ice-covered ocean can also be provided by these satellite data. The sea ice information includes; ice edge location, ice concentration, floe size distributions, ice type, and ice kinematics. The merging of remote sensing data with acoustical data can potentially provide new insight into understanding the interior structure of the ocean. The use of acoustical data to aid in the interpretation of remote sensing oceanographic data can in turn increase the use of the satellite information.

8:55

NN2. Satellite impact on environmental acoustics. S. A. Piacsek, F. Jensen, and P. Van Meurs (SACLANT ASW Research Center, I-19026, La Spezia, Italy)

The continuous surveillance of atmospheric and oceanic conditions at the sea surface by remote sensing platforms can be exploited to improve acoustic propagation prediction. This improvement is especially important in areas where surface observations are not usually taken or are difficult to take, e.g., in oceanic areas off the main shipping routes or under stormy weather patterns. It has been recently shown that both the climatology of the water mass, as well as the analyzed and/or predicted atmospheric surface fluxes can contain significant errors that are larger than those associated with the remotely sensed signals. The purpose of this paper is to illustrate quantitatively the improvement in the prediction of the acoustic environment and propagation due to remote sensing in such situations by three examples. The first two will use a 1-D mixed-layer model coupled to a range-independent acoustic model to study the impact of joint scatterometer/IR observations on propagation in surface ducts. Both a situation where mechanical stirring due to the wind or where convective stirring due to strong surface cooling is dominant are investigated. The third example will study the improvement in the description of the vertical structure and the horizontal position of an oceanic front, and the corresponding acoustic propagation patterns, due to combined altimeter/IR observations. This study will use a 2-D ocean model coupled to a range-dependent acoustic model.

9:20

NN3. Ocean sensing and modeling as a basis for acoustic prediction. Jerald W. Caruthers, Jim L. Mitchell, Theodore J. Bennett, Jr., George E. Kerr, Paul W. May (Naval Ocean Research and Development Activity, NSTL, MS 39529), and P. W. deWitt (Naval Oceanographic Office, NSTL, MS 39529)

Several ongoing projects at the Naval Ocean Research and Development Activity (NORDA) form a vertical program for the development of ocean sensing and modeling aimed at supporting acoustic predictions. These projects cover the spectrum from basic research to advanced development. This paper reviews an emerging philosophy for the integration of these several disciplines and presents an example of the application of a part of that philosophy. The example provides comparisons among acoustic propagation runs for various representations of a 600-km-long, vertical section across the Gulf Stream in the northwest Atlantic Ocean. These representations include (1) a dataset of 700-m XBT's merged with historic profiles to the ocean bottom and spaced at 20 km; (2) an optimally interpolated set of profiles spaced at 5 km; (3) a dataset developed from altimetric data spaced at 5 km; and (4) a single point profile. [Work supported by NORDA, ONT, and the Naval Oceanography Program.]

9:45

NN4. Ocean acoustic tomography and acoustic propagation. R. C. Spindel and P. F. Worcester (Woods Hole Oceanographic Institution, Woods Hole, MA 02543 and Scripps Institution of Oceanography, La Jolla, CA 92093)

Ocean acoustic tomography is a method for remotely sensing deterministic and stochastic variations in the ocean sound speed field. The fundamental tomographic measurement is sound speed, and therefore tomography directly provides acoustic propagation models with this essential, first-order, environmental information. In one form or another the various sound propagation codes rely on sound speed data in either 2-, 3- or 4-dimensional form. Experiments have shown that both mean and range-varying sound speed data can be obtained using tomography, and that when appropriate assumptions are made about the known horizontal and vertical covariance structure of the ocean, these data are spatially continuous. Thus it is not necessary to choose an interpolation scheme for discrete profiles that may be mathematically sensible, but physically less meaningful. Further, since the tomographic measurement itself uses propagating acoustic waves, it affords an opportunity to obtain simultaneously selected sound propagation parameters. We discuss these and other aspects of tomographic remote sensing and its application to acoustic propagation.

Contributed Papers

10:10

NN5. Multiple receivers in single vertical slice ocean acoustic tomography experiments. Bruce M. Howe (Scripps Institution of Oceanography, University of California at San Diego, La Jolla, CA 92093)

Will additional receivers widely spaced in the vertical on each mooring of an (x,z) tomographic experiment improve the range-dependent estimates of the sound speed and water velocity fields? Numerical simulations were performed to answer this question. For a 300-km range, additional receivers reduce the error variance of the estimated perturbation sound-speed field $\delta c(x,z)$ a factor of 3 (as compared with the case with single hydrophones) to 3% of the *a priori* variance (0.5 m/s or $\sim$0.1°C rms). For a 1000-km range, range-dependent features are only marginally resolved, even with additional receivers and optimistic data error levels. Range dependence in the velocity $u(x,z)$ is unresolved in all cases.

10:25

NN6. Distributed acoustical assimilation in the equatorial Pacific Ocean. Thomas L. Clarke and John R. Proni (Ocean Acoustics Division, AOML/NOAA, 4301 Rickenbacker Causeway, Miami, FL 33149)

The possibility of using a dynamical-model-based equatorial ocean data assimilation technique as the basis for tomographic inversion is explored. In the equatorial Pacific relatively few ray paths come near the surface causing difficulty for most inversion techniques. A data assimilation technique being developed at the Physical Oceanography Division of AOML/NOAA [Thacker and Long, in preparation] uses a dynamical approach that can provide coupling between the deep conditions that are directly sensed by the acoustics, and the near surface events that are of direct climatic significance. The model decomposes temperature variability into dynamical modes identifiable with Kelvin and Rossby waves; the sensitivity of individual ray-path travel times to the modes is used as the basis for the inversion. The acoustic data is assimilated with other ocean measurements by using a LaGrange multiplier approach constrained by the dynamics. Ray tracing results using equatorial Pacific temperature data are presented which show the promise of dynamical-model assimilation based tomographic inversion as a means of detecting near surface temperature anomalies in the equatorial Pacific.

10:40

NN7. Ocean and satellite data sets for acoustical modeling. Jim L. Mitchell, Zachariah R. Hallock, and William J. Teague (Naval Ocean Research and Development Activity, National Space Technology Laboratories, MS 39529)

Quasisynoptic ocean data sets consisting of thermal sections of data collected from deep (800-m) Airborne Expendable Bathythermographs (AXBTs) and sea surface topography measured by overflights of the U. S. Navy GEOSAT altimetric satellite are presented and analyzed. These data, collected as part of the NW Atlantic Regional Energetics Experiment (REX) during August 1985, cover a significant portion of the Gulf Stream front from approximately 57°W to 68°W including three warm meanders and the intervening two cold meanders. Additionally, two warm core rings and one Sargasso cold core ring are included in the region mapped during the six-day period of the AXBT survey. Using climatological regression between the dynamic thickness (0/3000 dbar) and isotherm depth, these *in situ* thermal data are compared with sea surface topography as measured by the GEOSAT altimeter. These data sets form the basis of the range-dependent acoustical model runs discussed in detail in a jointly submitted invited paper NN3 in this session.

10:55

NN8. Estimating sea surface spectra with acoustic tomography. James H. Miller (MIT/WHOI Joint Program in Oceanographic Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543), Ching-Sang Chiu, and James F. Lynch (Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A technique for estimating space- and time-varying sea surface spectra using acoustic tomography is described. The technique uses acoustic (mode and/or ray) phase or travel time perturbations as data for the inversions. The inverse problems for spatially homogeneous and spatially nonhomogeneous frequency-directional wave spectra are discussed. Resolution and accuracy of the technique are addressed. Results of inversions of synthetic data are presented as well as an application of this technique to data taken during the MIZEX '84 preliminary tomography experiment. Directions of future research are indicated.

11:10

NN9. Sound-speed profile inversion in the ocean. Linda Boden and John A. DeSanto (Center for Wave Phenomena, Mathematics Department, Colorado School of Mines, Golden, CO 80401)

For sound-speed profiles which vary only in depth, a method has been developed whereby the scattered field data in frequency (or k-space) can be related to the sound-speed profile correction (from an assumed profile guess input) as a quasi-Fourier transform pair. The inversion is straightforward using this Fourier relationship. The method uses a Fourier–Bessel representation of the acoustic field, a Born approximation on the depth dependent part of the Green's function, a WKB representation for the wave functions and asymptotics and linearizations to derive the transform pair. In spite of all the approximations, good results are obtained for profile recovery using synthetically generated data.

NN10. Application of discrete linearized inversion to the sofar inverse problem. Peter Kaczkowski and John A. DeSanto (Center for Wave Phenomena, Mathematics Department, Colorado School of Mines, Golden, CO 80401)

Inversion for the one-dimensional sound-speed profile in the sofar waveguide is performed using damped least squares (Marquardt–Levenberg algorithm). The study uses the split-step parabolic equation method to solve for the synthetic acoustic pressure field in the ocean. Data for the inversion are also provided by the same model, and the robustness of this technique is investigated using different samples of the sound field: horizontal and vertical arrays of varying length and, single point data of varying bandwidth. The parameter resolution matrix and the singular value decomposition of the matrix of partial derivatives lend insight into the inversion and provide quantitative measures of the relative merit of different experimental configurations, and this will be discussed.

11:40

NN11. High-resolution time of arrival estimation via linear prediction. Ivars P. Kirsteins (Surface Ship Sonar Department, Naval Underwater Systems Center, New London, CT 06320)

A new method is presented for estimating the arrival times of overlapping signal pulses that are separated by less than the duration of the signal autocorrelation function. The method is based on the observation that if the signal has a flat band-limited spectrum, then the maximum-likelihood estimator for the time of arrivals can be approximately transformed into an equivalent high-resolution exponential parameter estimation problem. Then, an improved linear prediction algorithm [D. W. Tufts and R. Kumaresan, Proc. IEEE 70, 975–989 (1982)] for high resolution exponential parameter estimation is used to estimate the time of arrivals. The proposed method is computer simulated for two linear FM sweep signal pulses at various separations and signal-to-noise ratios. The simulation results are compared to the Cramer–Rao lower bound (CRLB). These results indicate performance close to the CRLB over a reasonable range of pulse separation and signal-to-noise ratios.

THURSDAY AFTERNOON, 11 DECEMBER 1986 SALON K, 1:30 TO 5:00 P.M.

Joint Meeting of Accredited Standards Committees S1 and S3

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

Meeting of Accredited Standards Committee S1 on Acoustics

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

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

Meeting of Accredited Standards Committee S3 on Bioacoustics

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

Standards Committee 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 that 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.

Session OO. Architectural Acoustics VI and Noise VII: Outdoor-to-Indoor Sound Transmission, Part 2: Sound Insulation of Building Elements

George E. Winzer, Chairman
Winzer Associates, P.O. Box 5717, Rockville, Maryland 20855

Chairman's Introduction—1:30

Invited Papers

1:35

OO1. Community aircraft noise ordinance—Sound transmission considerations. David Braslau (David Braslau Associates, Inc., 1313 5th Street S.E., Suite 322, Minneapolis, MN 55414)

The Twin Cities Metropolitan Council has established aircraft noise compability guidelines for major and intermediate airports. The Council has prepared a "Model Ordinance" to help communities meet these guidelines. It has adopted a "noise zoning overlay" concept based upon a review of ordinances from other communities in the U.S. and Europe. Interior noise levels were established during a one-day workshop in 1983; 45 dB was established as the A-weighted interior residential level, to be measured in the same units as exterior noise levels (L_{eq} or L_{dn}). Compliance can be achieved through the use of specified STC values for building elements or through analysis by a "recognized acoustical specialist," defined in the ordinance. Minimum required STC values were based upon the noise reduction needed for each land use category and noise impact zone as well as assumed standard values of relative building component area and associated STC, aircraft takeoff noise spectrum, angle of incidence, number of exterior walls, and room absorption coefficient. An exterior wall performance methodology and worksheet to assist in evaluation was developed and implemented on a LOTUS 1-2-3 worksheet. Noise reduction benefits of the State energy code were also evaluated. [Work supported in part by the Twin Cities Metropolitan Council.]

2:00

OO2. Exterior wall rating (EWR), a single number index for rating the sound transmission loss of A-weighted sound levels for exterior facades. Louis C. Sutherland (Wyle Laboratories, 128 Maryland Street, El Segundo, CA 90245) and Gary Mange (Western Electro-Acoustic Laboratory, Inc., 1711 Sixteenth Street, Santa Monica, CA 90404)

A single number index, called the exterior wall rating, or EWR, has been developed for rating the sound transmission loss of A-weighted sound levels for exterior facades. This index is closely patterned after, but distinctly different from, the STC single number index. The latter provides a reliable rating for the sound transmission loss of interior partitions for typical interior sounds, but is not suitable for assessing the sound transmission loss of exterior walls to major sources of outdoor noise. However, by applying principles similar to those upon which the STC rating was developed, a practical index was constructed which can reliably rate exterior walls in terms of their ability to attenuate A-weighted sound levels. EWR originally evolved out of a systematic analysis of 1/3-octave-band sound transmission loss data for over 225 wall constructions and 33 different window constructions with four different relative areas from 0%–20% of the wall areas. From this analysis of over 22 500 combinations of structural assemblies, EWR was found to be a substantially more accurate predictor of attenuation of A-weighted sound levels than other candidate indices. These included a modified STC, SIL, and average TL values for 16 1/3-octave bands. EWR includes a necessary correction for the different spectra of aircraft and highway noise, and thus provides an efficient method for the design or analysis of sound insulation treatment of buildings to attenuate noise from these major outdoor noise sources. One such application is considered in paper HH4 by David Brown [J. Acoust. Soc. Am. Suppl. 1 80, S67 (1986)]. The basic methods for calculating EWR values from standard TL data are defined to encourage sound-transmission-loss-testing-laboratory operators and users to apply this practical method for rating sound transmission loss of exterior walls.

2:25

OO3. Residential noise insulation for mild climates. Russell B. DuPree (Charles M. Salter Associates, 930 Montgomery Street, San Francisco, CA 94133)

Since 1974 the California Building Standards Law has required that new multifamily housing, hotels, and motels be designed to prevent the intrusion of exterior noise above a community noise equivalent level of 45 dB. At about the same time, the State also adopted energy insulation standards which require low air infiltration rates for new dwellings. While these requirements seem to be complementary from the standpoint of noise insulation, the relatively mild climate of costal communities often obviates the need for air conditioning or even mechanical ventilation. This paper discusses some of the design, cost, and enforcement conflicts between adequate ventilation and noise insulation for areas with mild climates.

2:50

OO4. Exterior sound transmission through fenestration products—Do we need a new rating method? Gregory C. Tocci (Cavanaugh Tocci Associates, Inc., Natick, MA 01760)

The California Association of Window Manufacturers (CAWM) requested ASTM to develop test criteria and performance ratings for exterior sound control capabilities of fenestration products. This presentation considers the relationship between sound transmission class (STC) and traffic, aircraft, and rail noise reduction (NR) for a group of glass configurations tested by Monsanto. It will be shown that a clear relationship exists between glass STC and noise reduction, which depends upon source spectrum shape and glazing configuration. It will be shown that any new rating method would necessarily take the form of the glazing STC rating plus a constant related to source spectrum type. Other features of building exterior construction will be considered, as they affect the minimum acceptable NR required for fenestration in any particular application. This obviates any benefit that a new rating scheme, dealing more directly with fenestration products, might have over the current STC rating method. A simplified method for estimating minimum required STC will be presented which will provide the advantage that the CAWM now seeks through a new fenestration products sound transmission loss rating. [Work partly supported by Monsanto.]

3:15

OO5. Window design techniques for sound attentuation. Thomas S. Stark (DeVAC, Inc., 10130 Highway 55, Minneapolis, MN 55441)

Excellent sound attenuation can be obtained with high performance, double window construction methods by optimizing glass selection and addressing basic design and construction details. With careful attention to manufacturing standards, DeVAC Series #600 thermo-barrier acoustical windows have STCs in the 37–39 range, glazed with common 1/8-in. window glass at the exterior and interior. The design provides an air space separation of 2-in. between inner and outer lites. Isolation and sealing for thermal performance increases the acoustical performance without using laminated glass. Noise penetration is reduced by achieving a low rate of air infiltration with dense, double weatherstripping, and covering drain holes with valve flaps to exhaust water buildup and eliminate air path channels. Glazing with heavier, unequal thicknesses of glass (maximum 1/4-in.) increases STC to 48 dB. Further options that accept glass up to 1/2-in. thick, and increase the air space separation to as much as 8-in., enable incremental improvements in STC to a currently tested 56 dB. The windows install in new construction with standard installation methods and, with optional trim accessories, adapt to existing openings for retrofit. Acoustical performance of available glazing and frame configurations are documented by sound loss test reports. Case history applications review both new construction and replacement projects requiring sound control windows which are operable for emergency ventilation or life safety, thereby addressing points of concern for residential, lodging, and educational facilities. Field testing by independent acoustical consultants confirms that the acoustical performance of installed windows meet project specifications.

Contributed Papers

3:40

OO6. Sound transmission through windows of high-rise buildings located adjacent to freeways and major traffic arterials. Jerry P. Christoff (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

A mathematical theory was developed for sound transmission through single glazed windows assuming a line sound source and the appropriate angles between the location of the window in the building and the source. The predicted internal sound pressure levels using the theory and sound transmission loss values from ASTM E90 are compared to measurements in several buildings.

3:55

OO7. Vertical pump noise control enclosure. Bonard E. Morse (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

When water districts began changing pumping equipment, replacing submersibile pumps with vertical pumps, they discovered a significant increase in the noise radiated by their equipment. This paper describes and discusses an enclosure that reduces the radiated noise and provides ducted air in and out for cooling the electrical drive motor. Data will be presented showing the noise levels before and after installation of the enclosure both near and at several distances from the pump. In addition to noise reduction the enclosure results in cooler operating temperatures which in turn aids in extending the service life of the pumping equipment.

4:10

OO8. Transmission loss of double panels with different gases in the gap. Mahabir S. Atwal (Paul S. Veneklasen and Associates, 1711 Sixteenth Street, Santa Monica, CA 90404)

Recently there has been a renewed and growing interest in increasing the transmission loss of structures without significantly increasing the mass of the structure. A common method of increasing the transmission loss of a panel is by the introduction of one or more additional panels with intervening air spaces. Such a multiple layer panel is much more effective than a single panel of equivalent mass. This paper describes the possibility of increasing the transmission loss of a double panel by using different gases in the gap. Experimental results are presented which show the effect on the transmissions of different gases in the gap. Results of preliminary theoretical analysis to predict the transmission loss of such double wall panels is also presented.

4:25

OO9. Sound absorption and transmission by periodically supported double panel structures. J. S. Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907) and N. C. Baines (Stewart–Hughes Ltd., Chilworth Manor, Chilworth, Southampton, SO9 1XB England)

Double panel structures are frequently used in modern lightweight constructions, e.g., as factory roofing. Measurements indicate that these structures may provide the majority of low-frequency absorption in the spaces they enclose. Conventional infinite double panel models significantly underestimate the absorption provided by these structures. As a result the authors have instead modeled these surfaces as two parallel infinite arrays of simply supported rectangular panels separated by a finite depth airspace, thus introducing component panel dimension as a model parameter. When the incident wave field is plane, the panel motions and the reflected, interpanel and transmitted acoustic fields may be expanded in functions periodic in the panel dimensions and their component strengths determined by application of sets of boundary conditions. The fraction of the incident energy that is either dissipated by each panel layer or transmitted may then be calculated. Absorption coefficients calculated in this way are consistent with those measured for factory roof structures and indicate that panel damping may control low-frequency sound absorption.

4:40

OO10. Noise transmission through building corridors and staircases in apartment houses. Effects on noise control of apartments. Alexandra G. Sotiropoulou (Technical University of Athens, Athens, Greece)

A series of noise attenuation measurements was carried out in long corridors and staircases at two, six-storied apartment houses in Athens. Such spaces are often liable to considerable transmission of building noises into apartments, due to their tubelike shape and their nonabsorptive treatment. Test spaces were treated in ordinary plaster; floors and stairs were covered in marble. Attenuation of 0 dB/m was measured in all corridors at all frequencies. In staircases, noise linearly decreased away from the source; the mean frequency attenuation was 5.5 dB per story. The effect of these findings within apartments was then evaluated, considering slamming doors, raised voices, etc. operating in builing corridors. In these evaluations, the sound insulation of typical entry doors of apartments was taken from measurements (STC 14). In all cases NR values exceeded the criterion value NR 35 [J. T. Broch, *Acoustic Noise Measurements* (B&K, Denmark, 1971), 2nd ed., Chap. 3, 35–39]. These results show that ordinary construction of corridors, staircases, and apartment doors is inadequate to protect apartments against usual building noises, and that construction improvements are required.

THURSDAY AFTERNOON, 11 DECEMBER 1986 — SALONS A AND B, 2:00 TO 5:00 P.M.

Session PP. Musical Acoustics VI: General Topics

William J. Strong, Chairman
Department of Physics and Astronomy, Brigham Young University, Provo, Utah 84602

Invited Paper

2:00

PP1. Residue pitch and beats as heard in a musical context. John R. Pierce (CCRMA-Music, Stanford University, Stanford, CA 94305)

Published residue pitch measurements give the pitch of the missing fundamental for tones consisting of three, and even two, successive harmonics over a wide range of harmonic numbers and frequencies. Presentations closer to a musical context give other results. Scales spanning an octave played with two and three sinusoidal components of constant frequency difference sound like steadily ascending scales. The last tone (frequency ratio 3:2 or 4:3:2) sounds higher in pitch than the first (frequency ratio 2:1 or 3:2:1) when played immediately after it. Other examples contrast the first 6 equal-amplitude harmonic partials with tones consisting of the upper 2, 3, 4, and 5 harmonics. These latter tones do not match the pitch of the 6-partial tone when the literature says they should. Tunes and scales were played with tones with two sinusoidal components spaced by the pitch frequency. The tune or scale is heard only at high levels, and "in the ear," not coming from the speakers. Can there be pitch without place? Reported residue pitches do not correspond to pitches heard in a musical context.

2:30

PP2. Reaction time and musical expectancy: Priming of chords with no partials in common. Jamshed Jay Bharucha and Keiko Stoeckig (Department of Psychology, Dartmouth College, Hanover, NH 03755)

Subjects were presented with two musical chords in succession, called the prime and target, respectively. The prime and target were either related (e.g., C and D major, respectively, sharing a parent key) or unrelated (e.g., C and F# major, respectively, sharing no parent key). Subjects judged, as quickly as possible, whether the target was in tune or out of tune. Response times for in-tune targets were faster when prime and target were related than when they were unrelated, suggesting that the prime generated expectancies for related targets. Priming occurred even when prime and target shared no partials. These results suggest that chordal expectancies generated by a musical context are not due solely to priming at the level of individual frequencies, but also involve priming at a more abstract level of chord function, implicating a cognitive representation of chord relationships. A network model of chord relationships is proposed whereby chord nodes activate related chord nodes via links to their parent key nodes.

2:45

PP3. A new score input method for computer music. R. Shyu (AT&T Bell Laboratories, Murray Hill, NJ 07974)

To reduce the time and effort of score inputting and score proofreading for computer music, a new score input method was designed, implemented, and investigated. This new method allows scores to be entered through a regular computer keyboard using the minimum possible number of keystrokes, and be displayed, with the aid of computer graphics on a computer monitor in original musical notations. The number of keystrokes is minimized using the idea of "music abstraction." An often used musical idea, no matter how many notes it consists of, can be abstracted as a single icon—called the "musicon" (music icon), and input using one keystroke each time it is needed. A study evaluated this design in terms of the speed of input and proofreading. The musicon can not only save keystrokes, it can also become a compositional tool by helping the composer to organize musical ideas in the same way a thought processor like the Thinktank can help a writer.

3:00

PP4. Acoustic parameters of violin vibrato. Howard B. Rothman and A. Antonio Arrayo (IASCP, ASB-51, University of Florida, Gainesville, FL 32611)

Studies of vocal vibrato conducted in 1930 indicated that the great singers produced a pulsation rate ranging from 5.9–7.8 pulses per second (pps). Studies conducted during the last decade show singers' vibrato pulsations ranging from 4.9–6 pps. The slower pulsation rate of modern singers reflects a different aesthetic that prevails today. Studies of violinists' vibrato conducted in 1930 indicate a pulsation range of 5.5–7 pps. Other findings from early studies indicate that violin vibrato rate remains constant as tonal amplitude varies and vibrato amplitude changes with tonal intensity. The present paper will examine the relation between the vibrato of violinist's representing an earlier aesthetic and those of today. Samples of sustained violin tones were obtained from acoustic, electric, and stereo recordings of great violinists of the past and the present. The samples were digitized using a 16-bit A/D converter at a sampling frequency of 10 kHz. Each digitized sample will be converted to a useful format for making purposes in order to derive information on vibrato pulse rate, the mean frequency of the tone, the semitone deviation around the mean, percent frequency deviation and percent amplitude deviation around the mean amplitude.

3:15

PP5. Pedagogy of tenor high voice mechanisms. John W. Large (Department of Music, Texas Christian University, Fort Worth, TX 76129)

The Italian pedagogical model was utilized in a study of tenor registration, focusing primarily on high voice mechanisms. Charts were developed to contrast the Italian model with the German model and to relate variable transition tones to five tenor classifications, five vowels, and three levels of intensity. Vocalizes designed to establish the Italian model were then taught to ten tenors, ranging from beginner to professional, with varying degrees of success. The model and its variables, the vocalizes, and the results, suggest that one transition and its resultant register are determined primarily by *resonance adjustment* and another transition and its resultant register by *laryngeal adjustment*. Examples of appropriate repertoire for each tenor classification and for each student subject are given.

3:30

PP6. Acoustical problems in the bassoon. John Backus (Physics Department, University of Southern California, Los Angeles, CA 90089-0484)

The bassoon is an orchestral instrument of considerable antiquity, which has evolved into its present form through centuries of tinkering. Unfortunately, it still has a number of deficiencies. Some notes are considerably out of tune. The played scale is not uniform; certain notes have a different quality from their neighbors. As an amateur bassoonist for some years, the author is well acquainted with these deficiencies, and hopes to correct them, using the acoustical knowledge and apparatus now available. To that end equipment has been set up to measure the resonance frequencies of the instrument, compare them with the playing frequencies, and ultimately to see what alterations in tone-hole size and location might bring about better intonation. The first measurements, on two instruments, show that the resonance frequencies are about 1/4 semitone below the playing frequencies for about the first octave, after which the resonance frequencies rather suddenly rise so as to be as much as a semitone above the playing frequencies. The reason for this behavior needs to be determined. The bassoon reed is a very important part of the instrument, and is the subject of much folklore. Some preliminary measurements on reeds indicate that much of this folklore may have no basis in fact.

3:45

PP7. Simulation of a player–clarinet system. Scott D. Sommerfeldt and William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

A time-domain simulation model has been developed for investigating the player–clarinet system. The three components which constitute the simulation model consist of the "subreed" system, reed, and clarinet. The "subreed" system (lungs, bronchi, trachea, and vocal tract) is represented in terms of an analogous circuit model to obtain the "subreed" pressure. The reed is represented as a damped, driven, nonuniform bar. The clarinet is represented in terms of its input impedance impulse response. A convolution of the impulse response with the volume flow through the reed aperture determines the mouthpiece pressure. Use of the model is valid for both small- and large-amplitude oscillations. Many of the nonlinearities associated with the clarinet are incorporated in the model in a rather natural way. Several vocal tract configurations are investigated to determine the influence of the vocal tract on the clarinet tone.

4:00

PP8. Impulse responses for reentrant discontinuities in bores. Anthony S. Lee and R. Dean Ayers (Department of Physics and Astronomy, California State University, Long Beach, CA 90840)

We establish an optical analogy for the processing of spherical wavefronts in conical bore segments. A reentrant discontinuity acts like a converging lens or mirror to give outgoing waves a more positive vergence than incoming waves [M. P. Keating, Am. J. Phys. **43**, 766–769 (1975)]. Impulse responses for single interactions with such a discontinuity in-

clude exponentially growing wakes, which become unmanageable very quickly in digital convolutions. Geometrical considerations rule out a reentrant discontinuity existing in isolation, and its troublesome behavior is kept under control by multiple reflections involving another discontinuity of stronger, negative power. In closed systems this process shows a "sorcerer's apprentice" type of behavior, in which a temporary solution to the problem turns into the problem for the next cycle. Fortunately that problem becomes more deltalike as time progresses, and there is a tendency for it to be cancelled by a neighboring delta. In open systems the behavior is much tamer, with the extended response dying out very quickly. Our piecewise analytic results suggest some approximations which may be useful in digital convolutions.

4:15

PP9. Science in the service of the performing arts. Section I: Background. Paul S. Veneklasen (Paul S. Veneklasen Research Foundation, 1711 Sixteenth Street, Santa Monica, CA 90404)

Has orchestra performance reached its zenith? What can science offer for sustenance? The recognized limitations are exposed. The pioneering work of physicists is reviewed. The inspiring efforts of Leopold Stokowsky are paid special tribute and his innovative leadership toward orchestral balance and dynamic performance are stressed. Experiments with spatial factors in performance are described.

4:30

PP10. Science in the service of the performing arts. Section II: The physical factors. Paul S. Veneklasen (Paul S. Veneklasen Research Foundation, 1711 Sixteenth Street, Santa Monica, CA 90404)

The choral anomaly intrigues us into experiments that disclose clearer understanding of projection of sound from the stage. The relative acoustical power output of instruments is the primary factor. The directional characteristics of musical instruments play a dominant role. Precise acoustical model tests disclose the power of physical manipulation of acoustical factors. Proper placement of performers can combat the effects of body baffling and absorption on stage. Manipulation of the surfaces of the orchestra enclosure offers the acoustician great versatility for intersection balance.

4:45

PP11. Science in the service of the performing arts. Section III: Experiments with full orchestra and conclusions. Paul S. Veneklasen (Paul S. Veneklasen Research Foundation, 1711 Sixteenth Street, Santa Monica, CA 90404)

A unique series of tests with the Seattle Orchestra in the calibrated Seattle Auditorium was staged to experiment with section arrangement within the orchestra enclosure. New values for relative and absolute power output from instruments and sections as a function of dynamic playing level were derived. Results demonstrate that relative placement and elevation of sections can be used to greatly improve intersection balance and choral effects. A unique and strange stage arrangement emerges.

THURSDAY AFTERNOON, 11 DECEMBER 1986 — SALONS 1 AND 2, 1:00 TO 5:00 P.M.

Session QQ. Physical Acoustics VI: Sound Propagation in Inhomogeneous Media

Murray S. Korman, Chairman

Department of Physics, U.S. Naval Academy, Annapolis, Maryland 21402

Invited Papers

1:00

QQ1. Nonlinear scattering of crossed ultrasonic beams in the presence of turbulence—Intensity measurements versus turbulent mach number. Murray S. Korman (Department of Physics, U.S. Naval Academy, Annapolis, MD 21402)

An experimental scattering arrangement involving the nonlinear interaction of two mutually perpendicular intersecting beams (frequencies $f_1 = 2.1$ MHz and $f_2 = 1.9$ MHz), overlapping in a region of turbulence is investigated. In the absence of turbulence no radiated scattering intensity I_+ is observed at the combination frequency, $f_+ = 4$ MHz. However, in the presence of turbulence scattering is observed at f_+. This radiated component shows considerable amplitude modulation and exhibits spectral broadening. The experiment is performed in a 12 ft deep $\times$20 ft$\times$20 ft section of the U.S. Naval Academy Hydrodynamics Tow Tank facility. Two 2.54 cm-diam transducer units are pulsed at f_1 and f_2, respectively. They are both located 1 m from the interaction region X and form a plane with a 4 MHz receiving unit located (in cylindrical coordinates) 3 m from X at the angular position of 45° from the axis of both sending beams. A 4.81-cm-diam submerged nozzle is aimed perpendicular to the crossed beam plane and located 20 nozzle diameters above it. Measurements of the scattered intensity I_+ versus nozzle exit velocity U_0, show that $I_+ \propto U_0^{3.2}$ over the range 27–62 ft/s. U_0 can be related to turbulent fluctuations in the interaction volume. Theory and experiment, for I_+, lead to predictions of an average turbulent length scale dependence on Reynolds number. The sensitivity of detecting turbulence with this apparatus is discussed and measurements of spectral broadening versus U_0 will be presented. [Work supported by the Naval Academy Research Council.]

1:15

QQ2. Sound propagation and scattering in discrete random scatterers. Akira Ishimaru (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

Recent advances in the theory of sound propagation and scattering in discrete random scatterers are reviewed together with some examples and applications. We first discuss single scattering theory, transport theory, multiple scattering theory, and diffusion theory and their range of validity. Different theories are needed depending on whether the scatterers are tenuous or nontenuous. If the concentration is low (sparse), all the particles can be considered as independent scatterers, while pair correlation needs to be included if the concentration is high. If the particle sizes are much greater than a wavelength and the particle is tenuous, the small-angle approximation is applicable. The diffusion approximation is useful for small scattering particles. We also discuss amplitude and phase fluctuations and pulse propagation. If the concentrations vary in the propagation path and the medium is inhomogeneous, there are significant differences between the propagation characteristics of the plane wave and the spherical or beam wave. This also explains the so-called shower-curtain effect. Imaging through such discrete scatterers can be treated in terms of the modulation transfer function. Several experimental data are also presented.

1:40

QQ3. Measurements of random structure in the ocean and the earth by use of acoustic fluctuations. Stanley M. Flatté (Department of Physics, University of California, Santa Cruz, CA 95064)

High-frequency ($\gtrsim$ 1 cpd) variations in travel time of acoustic transmissions over ocean mesoscale distances are known to be dominated by the effects of internal wave displacements of the sound speed stratification (Flatté, 1983). A typical application involves 10-ms-wide pulses travelling over 300 km. It has been demonstrated that this type of data can be inverted for internal-wave strengths. Information can be obtained about range-averaged mean energy level and about energy distribution in the vertical. Data from the 1983 Tomography Experiment (Worcester, Spindel, and Howe, 1985), consisting of 41 days of transmissions between two moorings 300 km apart, have been used to observe the statistical field strength of internal waves (Flatté and Stoughton, 1986; Stoughton, Flatté, and Howe, 1986). It is shown that the accuracy of measurement of these variances is not strongly affected by the partially saturated nature of the sound propagation. Application of the path-integral method of wave propagation through random media to the analysis of seismic-wave propagation through the Earth's mantle is underway. Seismic waves with periods of 1 to 10 Hz received at 100-km-square arrays on the surface of the earth exhibit paradoxical wave-front properties involving small phase (or travel time) fluctuations combined with large amplitude fluctuations. A description of these observations will be given.

2:05

QQ4. Applications of acoustic propagation to remote sensing of inhomogeneities in the atmosphere and oceans. Edmund H. Brown (905 15th Street, Boulder, CO 80302)

The inverse problem in the propagation of sound through inhomogeneous fluids, that is, determining quantitative parameters characterizing the inhomogeneity fields from the changes in acoustic waves propagating through them, forms the basis for acoustic remote sensing. After a brief review of the development of remote sensing methods, recent advances—and problems—will be discussed, including measurement of velocity fields, and profiles of temperatures versus altitudes or depths. The possibilities of obtaining phase information and much larger scattering cross sections from fields of Rayleigh scatterers when their spatial correlations do not vanish (a topic under way in studies of radar sensing of clouds) may present opportunities for improved acoustic remote sensing.

2:30

QQ5. Two-scale solutions of the coherence equations in inhomogeneous random media. Mark J. Beran (School of Engineering, Tel Aviv University, Ramat Aviv, Israel)

The two-scale method of solution of the coherence equations in inhomogeneous random media and present typical solutions that have been obtained will be reviewed. The method was first used to study caustic corrections and the effect of a random medium on these corrections. Subsequently it was applied to the solution of the fourth-order coherence function which yields the scintillation index and correlation of intensities as special cases. Recently the sixth-order coherence function was treated in order to determine the next highest moment of the probability distribution of the intensity fluctuations. Although most of the two-scale solutions for the scintillation index have been for plane wave and point-source initial conditions in a random medium without a background sound-speed profile, a solution for the scintillation index in a quadratic channel will also be presented.

2:55

QQ6. Geometric theories for propagating statistical moments. John J. McCoy (The Catholic University of America, Washington, DC 20064) and David H. Berman (Naval Research Laboratory, Washington, DC)

Ray-based models for propagating the statistical moments of a stochastic radiation field through an inhomogeneous, but deterministic, medium will be considered. The source of stochastic fluctuations in the radiation field in this case could be in the physical source itself, or in the interaction of an initially deterministic signal with a randomly rough boundary surface. The reradiated field in the latter case requires a stochastic description. Detailed calculations are considered for the second moment (signal coherence) and the fourth moment (intensity fluctuations and correlations) but the road to extensions to higher-order moments is discussed. The geometric theories are formulated as conservation laws written on certain Fourier transforms of the statistical moments. On the level of the two-point moment the Fourier transform is taken with respect to the difference coordinate, for the averaged coordinate held fixed. The form of the conservation law for wide angle, i.e., Helmholtz theory, propagation is presented as is the simplification obtained for narrow angle, i.e., parabolic wave theory, propagation. The question of caustic corrections in the context of a geometric theory of coherence is discussed. On the level of the four-point moment Fourier transforms are taken with respect to two of the Tartarski coordinates for locating the four points. Again the form of the conservation law for wide angle propagation is obtained first, and subsequent simplification introduced corresponding to the narrow angle approximation. The question as to when a complete theory can be formulated in terms of less than complete four-point information specified on a source plane is discussed. Incorporating a stochastic volume scatter mechanism into the basic formulisms is considered.

3:20

QQ7. Universal power spectra for acoustic turbulence: Applications to wind waves, $1/f$ noise, and classical second sound. Seth Putterman, A. Larraza, and P. H. Roberts (Physics/Math Department, UCLA, Los Angeles, CA 90024)

A continuum pumped full of wave energy at an amplitude sufficiently large so that reversible nonlinearities dominate irreversible linear response becomes wave turbulent. In the limit of high nonlinearity acoustic turbulence and wind wave turbulence accumulate at $1/f$ and $1/f^5$ power spectra, respectively. A wave turbulent system can support new propagating energy modes analogous to second sound in superfluid He^4. This hyperbolic (nondiffusive) transport could account for the anomalous diffusivity observed in plasma devices and for the difficulties faced in achieving confinement. The key to the understanding of these phenomena is the nonlinearity in the continuum mechanics which leads to three basic effects: (1) scattering of sound by sound to produce waves with sum and difference frequencies; (2) refraction of waves by a slowly varying (inhomogeneous) background; (3) reaction of the background due to changes in the distribution of sound waves. Details of these processes are presented in the framework of the Euler equations.

Contributed Papers

3:45

QQ8. Prediction of the surface impedance of depth-varying ground surfaces. A. Berry and J. Nicolas (Mechanical Engineering Department, Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada)

Many ground surfaces have acoustical characteristics (density, porosity, flow resistivity) which vary with depth. Prediction of the surface impedance can be done by solving the wave equation in the nonhomogeneous medium. An exponential variation with depth of the parameters is usually assumed [R. J. Donato, "Impedance models for grass covered grounds," J. Acoust. Soc. Am. **61**, 1449–1452 (1977); K. Attenborough, "Acoustical impedance models for outdoor ground surfaces," J. Sound Vib. 99, 521–544 (1984)]. In this paper a new approach is presented. The surface impedance is calculated by discretizing the ground into a suitably large but finite number of homogeneous layers of known characteristics (multilayer approach). This method allows prediction for any depth variation of the ground parameters, and includes the simple hard-backing effect. The analytical results in terms of the impedance gradient, obtained by this multilayer approach are confirmed by theoretical manipulation of the fundamental equations describing the acoustic field in nonhomogeneous media. The numerical results show the typical tendancies caused by the ground inhomogeneity.

4:00

QQ9. A method for distinguishing between the attenuation due to scattering and the attenuation due to absorption for waves propagating in a dissipative, randomly inhomogeneous medium. Alan R. Wenzel (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

Waves propagating in real media such as the atmosphere, ocean, earth, seafloor sediments, etc., are subject to attenuation due to both intrinsic absorption and scattering by medium inhomogeneities. Measurements of attenuation in such media yield the combined effect of these two mechanisms but do not yield directly the individual effect of each of them acting in isolation. In order to obtain information on these individual attenuation mechanisms from measurements of the total attenuation, it is necessary to use analytical techniques based on the theory of wave propagation in the medium. The purpose of the present investigation is to derive such a technique for the case of scalar waves propagating in a one-dimensional, weakly dissipative, randomly inhomogeneous medium. The approach makes use of analytical expressions derived previously for the mean intensity and mean squared intensity of the field. With the aid of measurements of these two quantities, both the absorption coefficient and the scattering coefficient (of attenuation) of the medium can be obtained. [Research supported by NORDA.]

4:15

QQ10. Harmonic waves in a solid with periodically distributed inhomogeneities. J. D. Achenbach (Department of Civil Engineering, Northwestern University, Evanston, IL 60201)

A dispersion relation has been obtained for the propagation of harmonic waves in an elastic solid containing a three-dimensional array of regularly spaced inhomogeneities of identical shape. The analysis uses the reflection and transmission coefficients for a single layer of inhomogeneities. In this paper, a solid completely filled with periodically spaced inhomogeneities is considered as a stack of reflecting and transmitting planes. The dispersion equation can then be obtained on the basis of Floquet theory and by using the single-layer reflection and transmission coefficients. The results are valid in a specified frequency range, and provided that the layer spacing is sufficiently large so that exponentially decaying standing wave modes do not interfere with the neighboring layer. The general pattern of the frequency spectrum is one of passing and stopping bands. For spherical cavities and Griffith cracks, results are presented for the lowest acoustic mode (real-valued wavenumber), the transition to the first optical mode (imaginary wavenumber) and for a segment of the first optical mode (real-valued wavenumber) up to the first cutoff frequency.

4:30

QQ11. On the singularity index and the unfolding of the diffraction catastrophes. Michael G. Brown (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149-1098)

The caustics of high-frequency wave propagation may be classified using catastrophe theory. The wave field in the vicinity of any caustic is described by the corresponding diffraction catastrophe. The singularity index β is a measure of the rate at which such a wave field diverges as $\omega \to \infty$ at the point where all control parameters and moduli are set equal to zero. It is shown that away from this point β also describes a balance between two different measures of the unfolding of the wave field in each control direction, $\beta = \sigma_n \rho_n$. The indices σ_n and ρ_n describe, respectively, the rate at which individual ray arrivals separate in time and decay as a function of control parameter. Thus the behavior of the wave field in the vicinity of any caustic is governed to a large extent by a single number, the singularity index.

4:45

QQ12. Bubble swarm acoustics. Kerry W. Commander and Elan Moritz (Naval Coastal Systems Center, Code 4120, Panama City, FL 32407)

A study of the sensitivity of sound attenuation in bubbly liquids to variations of bubble distribution functions and void fractions was conducted using several theoretical models [A. L. Anderson and L. D. Hampton, J. Acoust. Soc. Am. **67**, 1865–1889 (1980); A. Prosperetti, Ultrasonics **22**, 115–124 (1984); R. Caflisch, M. J. Miksis, G. C. Papanicolaou, and L. Ting, J. Fluid Mech. **153**, 259–273 (1985)]. The distributions studied include uniform, Gaussian, naturally occurring, and others. The frequencies covered ranged from dc to 500 kHz. Differences in predicted attenuations between the theories are discussed as well as limitations of theories *vis-à-vis* empirical data. [Work supported by ONT.]

THURSDAY AFTERNOON, 11 DECEMBER 1986 — SALON F, 1:30 TO 4:45 P.M.

Session RR. Psychological and Physiological Acoustics VI: Pitch and Temporal Perception

John Preece, Chairman

Audiology Section, V. A. Medical Center, Long Beach, California 90822

Contributed Papers

1:30

RR1. The effect of tone location on temporal order discrimination. Kathy Barsz (Department of Psychology, SUNY, Stony Brook, NY 11794)

In order to examine the effect of tone location on temporal order discriminations, tones (200-ms onset–onset) in six-tone sequences were either presented all in one spatial location or were alternated between two spatial locations. On each trial a comparison sequence differed from the standard sequence, either such that the tones in each spatial location contained an order change or such that, when the tones were alternated between locations, only one spatial location contained an order change. In a 3IFC task, subjects chose the sequence in which the component tones were in a different order. Discrimination performance remained above chance in all conditions; however, performance was significantly worse when the tones alternated between locations and only one location contained an order change. These results support the hypothesis that temporal relationships between tones presented in the same spatial location are more salient than relationships between tones presented in different locations; they also support the assertion of Warren and Byrnes [Percept. Psychophys. **18**, 273–280 (1975)] that subjects can discriminate the order of perceptually unrelated tones better than chance, while extending the result of Bregman and Campbell [J. Exp. Psychol. **89**, 244–249 (1971)] that subjects perform worse when discriminating the order of perceptually unrelated tones than when discriminating the order of perceptually related tones.

1:45

RR2. Detection of a temporal fluctuation in the frequency ratio of two simultaneous tones. Laurent Demany (Laboratoire de Psychologie Expérimentale, CNRS LA 316, 28 rue Serpente, 75006 Paris, France) and Catherine Semal (Laboratoire de Psychologie Différentielle, CNRS UA 656, 28 rue Serpente, 75006 Paris, France)

Listeners were presented with two simultaneous frequency-modulated pure tones, each at 45 dB SPL. The two tones were heard dichotically and their carrier frequencies were separated by either 1 oct (1200 cents) or a slightly different interval (1200 ± 25, 50, or 100 cents). Each carrier frequency was sinusoidally modulated by $\pm 5\%$ at a rate of 2 Hz. Using a 2IFC procedure, we measured just-noticeable phase differences (jnpd's) between the two modulations. When the modulations were in phase, the two tones maintained a steady frequency ratio (the ratio of their carrier frequencies). Dephasing the modulations produced a quasisinusoidal frequency ratio fluctuation, which increased in amplitude with the phase difference. The results show that, for two tones with carrier frequencies below 2 kHz, jnpd's tend to be smaller when the interval formed by the carrier frequencies is equal or close to 1 oct than when it is ± 100 cents from 1 oct. This indicates that deviations from an octave interval may be easier to detect than deviations from an inharmonic interval even when monaural periodicity cues of the octave relationship are absent. However, the observed octave effects only occurred within a limited and listener-dependent frequency domain.

THURSDAY PM

2:00

RR3. Discrimination of five-component tonal complexes differing in crest factor. John P. Preece, Richard H. Wilson, and John T. Arcos (Audiology Section, V. A. Medical Center, Long Beach, CA 90822 and Division of Otolaryngology—Head and Neck Surgery, University of California, Irvine, CA 92717)

Hartman *et al.* [J. Acoust. Soc. Am. **79**, 1915–1925 (1986)] presented data on discrimination of tonal complexes differing only in number of spectral components within a fixed bandwidth. The Hartmann *et al.* data suggested that discrimination could be based upon perception of temporal fluctuations in the intensity levels of the sounds, or upon the detail of the spectral structure of the complexes. The present experiment assessed discrimination of tonal complexes that differed in only the phase relationships among five fixed frequency components. The phase relationships established a range of crest factors (peak/rms amplitude) for the complexes. When presented at equal loudness, the tonal complexes were easily discriminable by a musically trained subject, and were discriminable by all subjects with training. The data from the current study support a model of discrimination based only upon the temporal fluctuations in the complex envelopes. Discrimination between tonal complexes can be made for complexes differing either in rate or depth of such fluctuations, as long as the fluctuations are within the limits of temporal integration of the ear.

2:15

RR4. The temporal analysis of nonrepetitive auditory patterns. Donald A. Robin (Department of Speech Pathology and Audiology, The University of Iowa, Iowa City, IA 52242), Cecil W. Thomas, and Fred L. Royer (Case Western Reserve University, Cleveland, OH 44106)

Subjects were asked to listen to six-element auditory patterns comprised of 440-Hz pure-tone bursts separated by silent intervals. In a given pattern, two of the elements were closer in time (short ISI) than the others (long ISIs). Subjects were asked to report which of the two elements were closest. Tone durations for a given trial were 80, 60, 40, 20, or 10 ms. Long ISIs were 64, 32, 16, or 8 ms. Short ISIs were varying fractions of the long ISI. Performance decreased as either tone duration, long ISI, or the difference between long and short ISI decreased. Results suggested that there was a categorical response such that high accuracy ($>80\%$ correct) was maintained to some point and then performance markedly deteriorated. This cutoff point was at similar durations and intervals as found for the unitization of six-element repetitive patterns [F. L. Royer and D. A. Robin, Percept. Psychophys. 39, 9–18 (1986)]. Positions 2 and 4 were the most difficult, followed by 3, 1, and 5. Results are discussed with respect to a temporal frequency analyzer that operates like a bandpass filter, with the cutoff frequencies determining pattern smoothing and thereby subjects' ability to analyze a sequence.

2:30

RR5. Temporal specificity. Beverly Leis-Rossio and Arnold M. Small (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

The listener's response process is modeled as a temporal window through which the incoming stimulus waveform is viewed with the window positioned so as to maximize signal-to-noise ratio. To test this model, listeners were asked to detect a 0.5-ms pulse in a noise background during a 1500-ms observation interval. In a yes/no task, a primary (fixed temporal position) signal was presented on 40%, a probe (variable temporal position) on 10%, and noise alone on 50% of the trials. Listeners' performance, which was measured as hit rate, was maximal at those positions near the primary, independent of primary position. In companion experiments not utilizing the probe signal method, it was determined that signals positioned at any point within the observation interval were equally detectable if presented in isolation. However, if a signal could occur with equal probability at any position (temporal uncertainty), then performance worsened. This decrease can be accounted for by a temporal window which changes from a narrow configuration yielding a large signal-to-noise ratio in the first instance to a broadened state producing a reduced signal-to-noise ratio in the second case.

2:45

RR6. Detection of specific periods in a random sequence. Ervin R. Hafter and Lyne Plamondon (Department of Psychology, University of California, Berkeley, CA 94720)

Consider a hypothetical neural dimension in which is coded the period between peaks of an acoustic envelope. In hopes of better understanding the organization of this dimension, a kind of "period noise" is used to mask a "period signal." The masker is a sequence of high-frequency clicks spaced with periods chosen at random from the range 2–10 ms. The signal is a specific period added to the mask, replacing noise periods at randomly chosen locations in the sequence. Using a two-interval, forced choice, thresholds are defined as the proportion of "signals" in the sequence necessary for 71% correct detections. Signals tested ranged from 2–10 ms. In order to prevent spurious cues, the number of clicks in each interval is chosen to make the duration equal, and the levels are varied at random from interval to interval. Extensive data from two subjects show very different pictures of the relation between sensitivity and period. From one, thresholds increase (8%–29%) with signal period, as might be expected from studies of frequency discrimination; from the other, thresholds are reasonably constant (about 7%), regardless of the period tested. [Work supported by NINCDS.]

3:00

RR7. Detection of partially filled gaps. David M. Green and Timothy G. Forrest (Psychology Department, University of Florida, Gainesville, FL 32611)

Decrements in broadband noise stimuli were produced by changing the intensity of the noise for a brief interval. As the duration of the interval increases, less change in level is needed to detect the alteration. One can interpret these experiments as probing the impulse response of the auditory temporal mechanism in the manner suggested by Buunen and van Valkenburg [J. Acoust. Soc. Am. 65, 534–537 (1979)]. Unlike their results, our measurements are well fit by an exponential function with a short (8–10 ms) time constant. Our results are consistent with the model proposed by Viemeister [J. Acoust. Soc. Am. 66, 1364–1380 (1979)] to explain his results obtained with the modulation transfer function (MTF). Direct measurement of the modulation transfer function for these same listeners indicates good agreement between the time constant measured in the gap experiment and the low-pass break point measured in the MTF. We also explored increments in the noise stimuli instead of decrements. The data show symmetry when the increase or decrease in the noise rms is used to express the change in the stimulus. [Work supported by a grant from the National Science Foundation.]

3:15

RR8. Effect of frequency region on pitch discrimination. Virginia M. Richards and Ervin R. Hafter (Department of Psychology, University of California, Berkeley, CA 94720)

The ability to discriminate between pairs of signals having different "missing" fundamental frequencies was measured. The listeners' task was to distinguish a "standard" four-component complex from one composed of the same four harmonics, but a slightly different fundamental frequency. The complexes occupied one of three frequency regions. They were composed of the 3rd through 6th (low), 12th through 15th (moderate), and the 16th through 19th (high) harmonics of fundamental frequency. Several fundamentals were used, the median value being 250 Hz. Thresholds were determined as a function of the duration of the signal. Durations ranged from 20 to 190 ms. For both high and moderate frequency regions, discriminability improved steadily with the duration of the signal. For the low-frequency region, thresholds decreased rapidly as the duration was increased from 20 to 32 ms. However, for these signals, increasing the signal duration beyond 32 ms led to little additional improvement in performance. [Work supported by NIH and UC Chancellor's Patent Fund.]

3:30

RR9. Tone color and spectral spaces for steady sounds. Evi Papachristou, William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602), and Bruce L. Brown (Department of Psychology, Brigham Young University, Provo, UT 84602)

Perceptions of steady sounds are often described in terms of the attribute's pitch, loudness, and tone color (timbre). Tone color may be defined as that attribute by which a listener can judge two steady sounds having the same pitch and loudness to be dissimilar. Various researchers have sought underlying spectral spaces (physical) that would correlate well with tone color spaces (perceptual). For example, Klein *et al.* [J. Acoust. Soc. Am. **48**, 999 (1970)] studied vowel sounds and Plomp [Aspects of Tone Sensation **93** (1976)] studied musical sounds. For purposes of this study, twenty steady sounds (ten vowel and ten instrumental) equated for pitch and loudness were synthesized. Three spectral representations for each of the twenty sounds were used: (1) one-third octave-band spectra normalized to the overall sound level; (2) specific loudness per one-third octave band; and (3) F-weighted loudness per one-third octave band [S. S. Stevens, J. Acoust. Soc. Am. **51**, 575 (1972)]. Each of the spectral representations was subjected to a principal components analysis to reduce its dimensionality. The one-third octave spectra and loudnesses produced very similar results for the first four principal components, but the first four components of the F-weighted loudnesses accounted for less of the total variance than those of the other two representations. In separate analysis of the vowels and the instrumental sounds, the two were found to produce markedly different correlation matrices and spectral spaces. Listener tests using paired comparisons are currently in progress.

3:45

RR10. Frequency microstructure of pitch-level functions. E. M. Burns, K. E. Hoberg, R. S. Schlauch, J. Y. Bargones, and W. B. Beaman (Department of Speech and Hearing Sciences, JG-15, University of Washington, Seattle, WA 98195)

In a previous paper [E. M. Burns, J. Acoust. Soc. Am. Suppl. 1 77, S95 (1985)], the finding of no correlation between the frequency microstructure of the absolute threshold, associated with the presence of otoacoustic emissions, and the form of pitch-level functions measured at various points on the threshold microstructure was reported. In the experiments reported in this paper, pitch-level functions are measured at finer (20-Hz) frequency intervals. Cyclical changes in the magnitude and direction of the low-level portions (20–40 dB SPL) of the pitch-level functions that correlate strongly with the approximately cyclical changes in threshold microstructure are found. [Work supported by NINCDS.]

4:00

RR11. Auditory temporal summation in humans. George M. Gerken, Vishwa K. H. Bhat, Margaret Hutchison-Clutter, and Karen L. Donnelly (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

Monaural sets of thresholds were obtained for 11 human subjects with normal hearing using 24 digitally generated stimuli with cosine onsets and offsets. The purpose of the work was to evaluate modeling of temporal summation data, particularly with respect to the duration assigned to rise–fall time. Assuming similar central mechanisms, it should be possible to combine the temporal summation functions for single- and multiple-burst stimuli by selection of the duration assigned to rise–fall time. This duration was determined by regression analysis and was found to be, on the average, 39% of the *equivalent duration* calculated on the basis of stimulus power [P. J. Dallos and W. O. Olsen, J. Acoust. Soc. Am. **36**, 743–751 (1964)]. A combined summation function was obtained for each subject, and the average slope was 0.593. In no subject did the regression line account for less than 96.5% of the data variance. From these and other results in this experiment, it is concluded that the auditory system does not operate as a simple power integrator. [Work supported by NIH.]

4:15

RR12. An auditory paradox. Diana Deutsch (Department of Psychology, University of California—San Diego, La Jolla, CA 92093)

This paper describes and demonstrates a pattern of tones with some remarkable properties. When played in one key, it is heard as ascending; yet, when played in a different key, it is heard as descending instead. When a tape recording is made of the pattern and it is played back at different speeds, the pattern is heard either as ascending or as descending, depending on the speed of playback. To add to the paradox, when the pattern is played in any one key it is heard as ascending by some listeners but as descending by others. This effect is shown to be robust in the face of a number of parametric manipulations. Implications for processes underlying pitch perception are discussed.

4:30

RR13. Matching the pitch of a mistuned harmonic in an otherwise periodic complex tone. W. M. Hartmann (Department of Physics, Michigan State University, East Lansing, MI 48824), Stephen McAdams, and Bennett K. Smith (Institut de Recherche et Coordination Acoustique/Musique, 31, rue Saint-Merri, F-75004, Paris, France)

Listeners adjusted the frequency of a sine tone to match the pitch of a single mistuned harmonic in a complex tone consisting of 16 harmonics. Fundamental frequencies were 200, 400, or 800 Hz, and mistunings ranged from 0.5% to 8.0%. The successful matches, those close to the actual mistuned harmonic, show highly consistent and unexpected pitch shift effects; the pitch shifts can be twice as large as the frequency shifts, for both positive and negative frequency shifts. The percentage of matches which are successful provides an estimate of a listener's ability to segregate the mistuned component from the complex tone. Best performance occurs for mistuned harmonics in the region of spectral dominance. The highest *frequency* which can be segregated is a decreasing function of the fundamental frequency. The latter result suggests that the segregation operation depends upon both place and time coding in the auditory system. [Work supported by the NIH, the NSF, and the CHRS—France.]

Session SS. Shock and Vibration VI: Special Vibration Topics Including Damping

Harry Himelblau, Chairman
Jet Propulsion Laboratory, M.S. 150-300, 4800 Oak Grove Drive, Pasadena, California 91109

Contributed Papers

1:00

SS1. Vibration damping performance—What we should know about it. Pranab Saha (Blachford Engineers, P.C., 1899 Orchard Lake Road, Suite 105, Pontiac, MI 48053)

The vibration damping performance of materials that are used in the automotive industry are usually tested by either (1) the Geiger plate test method or (2) the complex modulus test method. The first test method has some advantages for testing materials that are unique to the automotive industry. Both of these methods have certain limitations, though the second method is a superior one in the sense that it can measure the damping performance with temperature and frequency. The limitations of some of the ways the data are typically presented using this method as well as how design charts can be used to overcome these limitations and compute the damping performance of the composite material (composite loss factor) for a given application are discussed.

1:15

SS2. Resonant transfer in longitudinal vibration testing. Isabel Jimeno-Fernandez, H. Überall (Department of Physics, Catholic University, Washington, DC 20064), W. M. Madigosky, and R. Fiorito (Naval Surface Weapons Center, Silver Spring, MD 20910)

Materials characterization can be carried out experimentally by measuring the complex modulus of linear viscoelastic materials from the displacement ratio of a bar, excited at one end, while the other end (which may carry an end mass) is allowed to move freely. This method, devised by Norris and Young, was recently developed [W. M. Madigosky and G. F. Lee, J. Acoust. Soc. Am. 73, 1374 (1983)] to cover a frequency range of 25–20 000 Hz. Standing waves cause resonances in the transfer function, and the positions and widths of these determine the real and imaginary parts of the modulus. The individual resonances are exhibited through a meromorphic representation of the transfer function; the corresponding facilitation in measuring resonance widths then leads to the possibility of experimentally determining the loss function of the material with improved accuracy.

1:30

SS3. Reverberant acceleration fields on a plate with both boundary and surface damping. C. E. Wallace (Department of Mechanical and Aerospace Engineering, Arizona State University, Tempe, AZ 85287)

Expressions for the direct and reverberant flexural vibration energy densities are derived for a thin plate excited by a random force applied at a point. The plate has both a surface and a boundary damping treatment. An expression for the vibration power introduced by the random force is presented and the maximum damping which will allow for an accurate measurement of the reverberant acceleration field is computed. [Work supported by Garrett Fluid Systems.]

1:45

SS4. Vibration power measurements using a damped plate as a reverberant structure. C. E. Wallace and Thomas Irvine (Department of Mechanical and Aerospace Engineering, Arizona State University, Tempe, AZ 85287)

A thin damped plate is used, in much the same way as a reverberant sound chamber, to measure the vibration power introduced by a shaker at a point on the plate surface. The power dissipated by the reverberant field is computed from the reverberant field acceleration measurements and compared with the power computed from impedance head measurements. These results are presented in a 500-Hz bandwidth for frequencies up to 4500 Hz. With the exception of the first 500-Hz band, the agreement is within 0.8 dB when the plate damping is determined for individual modes by the exponential method. When the damping is found from the decay of the reverberant field, the agreement is very poor. [Work supported by Garrett Fluid Systems.]

2:00

SS5. A solution for the response of a nonlinear oscillator subject to nonstationary stochastic excitation. Emilios K. Dimitriadis (Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061) and P. T. D. Spanos (Department of Mechanical Engineering and Materials Science, Rice University, Houston, TX 77251)

Envelope statistics are being frequently used to describe stochastic processes when a more detailed description is either unobtainable or impractical. The response of a nonlinear oscillator subject to nonstationary stochastic excitations is described. This is accomplished by analytically computing the transition probability density of a one-dimensional Markov process, which approximates the amplitude envelope of the response. For this purpose, an appropriate variation of the solution to the corresponding linear problem is employed. The probability density at various times as well as the mean-square and standard deviation of the amplitude envelope are computed for specific nonlinearly damped oscillators subject to separable and nonseparable nonstationary excitations. The analytical results are compared with those obtained by a Monte Carlo simulation study. The agreement between the two supports the applicability of the method to a broad class of oscillators.

2:15

SS6. Broadband generalized nearfield acoustical holography. Earl G. Williams (Naval Research Laboratory, Code 5133, Washington, DC 20375-5000) and Karl B. Washburn (Sachs/Freeman Associates, Inc., 1401 McCormick Drive, Landover, MD 20784)

Generalized nearfield acoustical holography (GENAH) has been applied to the study of the radiation from the vibration of cylinders radiating underwater. This experimental technique provides a complete description of the acoustic field from the surface of the vibrator to the farfield. Pressure, vector velocity, and vector intensity fields in three dimensions are reconstructed from a (two-dimensional) cylindrical hologram of the acoustic pressure measured in the nearfield of the source. One of the limitations of GENAH has been the use of a discrete excitation frequency, usually a resonance frequency of the cylinder structure. A new hologram had to be measured for each frequency of interest. We have eliminated this restriction by developing a broadband excitation technique, so that a single broadband hologram contains several octaves of information. The broad frequency band is derived from a one-cycle, modified sine wave, which powers a mechanical shaker driving the shell. Narrow-band holograms are then created from this broadband hologram by data reduction. Applying GENAH to these holograms provides a complete, three-dimen-

sional description of the radiation from the cylinder over several octaves. Also, the modes of radial vibration are obtained along with a three-dimensional mapping of the vector velocity and intensity fields. Experimental results will be shown.

2:30

SS7. Application of finite element methods to the prediction of power limitation in high-frequency transducers. Gerard Vanderborck, William Steichen, and Yves Lagier (Thomson-Sintra ASM, Chemin des Travails, 06802 Cagnes-Sur-Mer, France)

In the field of high-frequency acoustic arrays for high-resolution sonars, some transducers are driven with very high-power density and long pulse length. This produces a temperature rise in the transducer, leading to the variation of the materials characteristics. This phenomenon may reach the mechanical breaking point. To study this problem, we have developed a tridimensional piezoelectric finite element computer program which takes into account radiation (Helmholtz integral equation) and losses (mechanical, dielectric, piezoelectric). After an acoustic computation, using modal analysis, heat sources due to losses are determined in all the volume of the transducer. A finite element program computes the transient evolution of the temperature. According to the obtained temperatures, materials coefficients are updated and a new acoustic calculation is performed. The process is repeated until maximum dynamic stresses or maximum temperature is reached. Several examples are discussed including transducers with and without matching layer. [Work supported by DRET, France.]

THURSDAY AFTERNOON, 11 DECEMBER 1986 SALON E, 1:30 TO 5:06 P.M.

Session TT. Speech Communication VI: Coarticulation; Timing; Production Models

Edward P. Neuburg, Chairman
RSA, National Security Agency, Fort Meade, Maryland 20755

Contributed Papers

1:30

TT1. The "trough" phenomenon in Turkish and in English. Suzanne E. Boyce (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Previous studies on several Indo-European languages have shown that for strings of two rounded vowels separated by phonologically rounding-neutral consonants, both EMG and lip protrusion movement traces show double peaks coincident with the two rounded vowels plus an intervening "trough" [Bell-Berti and Harris, J. Acoust. Soc. Am. **71**, 449–454 (1982); Perkell, Speech Commun. **5**, 47–68]. This has been interpreted as due to independent rounding gestures associated with each vowel and as counterevidence to the claim that rounding coarticulation is anticipatory. In this paper, English speakers' production of nonsense words /kuktluk/, /kuktuk/, /kukuk/, /kutuk/, and /kuluk/ will be compared with that of speakers of Turkish, a language with strong constraints against the cooccurence of rounded and unrounded vowels in the same word. Both movement and EMG evidence will be used to argue that the trough phenomenon is more extensive in English than in Turkish. Implications for cross-language differences in coarticulatory patterns will be discussed. [Work supported by NIH.]

1:42

TT2. Vowel context, rate, and loudness effects on linguapalatal contact patterns in Hindi retroflex /ṭ/. R. Prakash Dixit (Department of Speech: Communication Disorders, Louisiana State University, Baton Rouge, LA 70803) and James E. Flege (Department of Biocommunication, University of Alabama at Birmingham, Birmingham, AL 35294)

Tongue position for retroflex /ṭ/ has been shown to differ across languages in the contexts of /a/ and certain other vowels. Degree of retroflexion has also been shown to be affected by vowel context [Ladefoged and Bhaskararao, J. Phonet. **11**, 291–302 (1983) and references therein]. Since previous studies have not determined whether the place or length of constriction differs as a function of vowel context, this study examined changes in the articulation of Hindi retroflex /ṭ/ as a function of vowel context, and also as a function of speaking rate and loudness. A 96-channel electropalatograph was used to monitor production of retroflex /ṭ/ in the nonsense words /bitíb/, /baṭáb/, and /butúb/ spoken in carrier phrase by a native speaker of Hindi. The length of constriction (i.e., the number of contacted rows × 2 mm) changed very little (< 1 mm) as a function of the three factors. However, the anterior–posterior location of the /ṭ/ constriction shifted dramatically as a function of vowel context. The center of constriction was 3.8 mm more posterior in /u/ than in /i/, and 5.8 mm more posterior in /a/ than in /i/. The pattern and area of tongue contact observed for retroflex /ṭ/ and dental /t/ were virtually identical in the /i/ but not in the /a/ or /u/ context. A similar vowel context effect on retroflex /ṭ/ was observed in speech produced at a fast rate, and in loud speech. [Work supported by NIH grant.]

1:54

TT3. The influence of phonetic context on the acoustical properties of stops. Mark A. Randolph and Victor W. Zue (Room 36-575, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

In American English, stop consonants may be released, unreleased, or deleted (e.g., the phoneme /t/ in "tea," "basketball," and "softly," respectively). The particular acoustical realization is almost obligatory in some environments and highly variable in others. The purpose of our study was to quantify the influence of context, including syllable structure, on the acoustical properties of stop consonants. Or database consisted of some 5200 stops collected from 1000 sentences. Phonemic transcriptions, including lexical stress and syllable markers, were provided and aligned with the speech waveforms. The stops were grouped into categories corresponding to their position within syllables (e.g., *syllable-initial-singleton*, *syllable-final-affix*, etc.) and marked according to their local phonemic context. Segment durations were measured and the stops were classified as released, unreleased, or deleted on the basis of their duration and voice onset time (VOT). In the analysis of these data, including the examination of VOT and other durational measurements, we found substantial effects of both syllable structure and phonemic context. For ex-

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ample, over 98% of the stops in syllable-initial position are released, regardless of the preceding phoneme. Syllable-final singleton stops are released 65% of the time when followed by a vowel (as in "oak is"), and unreleased 73% of the time when followed by a consonant (as in "sickness"). A stop is most likely to be deleted (over 40% of the time) when it is the last member of a syllable-final cluster containing an obstruent and followed by a syllable-initial consonant (as in "mostly"). An /s/ preceding a stop consonant, or a semivowel following, has a stronger influence on its VOT if it appears within the same syllable. Similarly, a singleton stop's voicing characteristic affects the duration of a preceding vowel differently, depending on whether the two segments are in the same syllable. Detailed results will be presented at the meeting. [Work supported by an AT&T Bell Laboratories Cooperative Research Fellowship and by DARPA.]

2:06

TT4. Postvocalic consonantal voicing and constancy of syllable duration. H. S. Gopal (Callier Center, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235) and A. K. Syrdal (AT&T Bell Laboratories, IH6C-320 Naperville-Wheaton Road, Naperville, IL 60566)

In American English, a vowel preceding a voiced consonant has frequently been shown to be of longer duration than one preceding a voiceless consonant. However, Port [J. Acoust. Soc. Am. 69, 262–274 (1981)] reported no effect of postvocalic consonant voicing on the total duration of the VC unit. Port attributed this finding to an inverse relationship between vowel and consonant durations. Our study replicates and extends this work to the entire CVC syllabic unit across changes in speaking rate. Syllable and segmental durations were measured for /pVC/ target syllables. The target syllables used eight vowels and voiced or voiceless cognates of the final consonant, which was either stop or fricative. Syllables were produced in sentence contexts at three speaking rates by four female and four male native American English speakers. [Work supported in part by NIH.]

2:18

TT5. Acoustic and articulatory evidence for consonant–vowel interactions. Carol Fowler (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Psychology Department, Dartmouth College, 209 Gerry Hall, Hanover, NH 03755), Kevin Munhall, Elliot Saltzman (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), and Sarah Hawkins (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Linguistics Department, Cambridge University, Sidgwick Avenue, Cambridge CB3 9DA, England)

Previous acoustical research [e.g., C. Fowler, Phonetica 38, 35–50 (1981)] has suggested that consonants and vowels are not produced strictly sequentially but rather overlap in time. By this account, the degree of co-production of underlying segments affects the pattern of the resulting acoustical durations. In the present experiment, two subjects produced utterances of the form CVC(C), where the vowels were either /ɛ/ or /æ/ and the following consonants were /p/, /k/, /s/, /ps/, /ks/, /sp/, or /sk/. Movements of the lips and jaw were monitored with a Selspot system. Laryngeal movements were measured by means of transillumination, and the accompanying acoustic signal was recorded. Analysis of the acoustic durations revealed that vowel duration was shorter when followed by a cluster than by a singleton in both vowel contexts. However, the following consonant durations were longer for the vowel /ɛ/ than for /æ/. Analyses of the lip and jaw kinematics suggested that the vowel shortening in the cluster contexts was produced in part by greater temporal overlap of the underlying segments. The influence of the intrinsic vowel duration on the following consonants may reflect a syllabic constraint on articulatory timing. The results will be discussed in terms of the articulatory structure of phonetic segments and the sequential coordination of consonant and vowel gestures. [Work supported by NINCDS.]

2:30

TT6. Spanish as a "syllable-timed" language. Mariscela Amador-Hernandez (Phonology Laboratory, Linguistics Department, University of California, Berkeley, CA 94720 and Speech Plus, Inc., 461 North Bernardo Avenue, Mountain View, CA 94043)

Spanish has been characterized as "syllable-timed" as opposed to English and German, which have been called "stress-timed" languages. This report describes a search for phonetic correlates of the term "syllable-timed." Measurements were made of the relative durations of the stressed and unstressed vowels in polysyllabic words spoken by four speakers of Mexican Spanish. A perceptual experiment was then conducted using the method of self-adjustment, where subjects could independently control the duration of each vowel in a synthesized polysyllabic word in order to produce stress on a certain specified syllable. Vowel durations were slightly different for the same test word in production and perception studies, with the unstressed vowels of the latter being more uniform. Although absolute equality of vowel (or syllable) duration was not found (since stressed vowels are always longer than unstressed), unstressed vowels generally had uniform durations. This uniform duration phenomenon appears to be what underlies the perception of "syllable-timed" in Spanish.

2:42

TT7. Rate-dependent variability in English and Japanese complex vowel $F2$ transitions. William B. Dolan and Yoko Mimori (Department of Linguistics, University of California, Los Angeles, CA 90024)

Are the effects of changes in speech rate on $F2$ transition slopes in languages with and without unitary diphthongs explicable in terms of general phonetic and physiological principles? Five American English diphthongs and four Japanese vowel sequences were recorded, spoken at three different tempos by eight and seven speakers, respectively. LPC spectra were computed at 10-ms intervals, and the resulting $F2$ contours were analyzed by a computer program designed to objectively define and measure steady-state and transitional portions of the vowel. Contrary to the claims of Gay [J. Acoust. Soc. Am. 44, 1570–1573 (1968)], speech rate was found to significantly affect two measures of transition slope in English, with slope decreasing as speech rate increased. Individual diphthong populations in both languages displayed high correlations between a transition, slope and its $F2$ range, indicating that rate adjustment is not simply a matter of target undershoot at faster speeds. Japanese showed less rate-dependent variability, suggesting that temporal reorganization for different speech rates is affected by language-specific structures. [Work supported by NIH.]

2:54

TT8. Artificial choral speech: Using digital signal processing algorithms in speech research. Edward P. Neuburg (National Security Agency, Fort Meade, MD 20755)

Much psychoacoustic speech research is done with nonspeech, such as synthetic speech or linear predictive coded (LPC) speech; such stimuli are repeatable and controllable, but not natural. Nondestructive alteration of natural speech to impose known, controllable parameters is becoming practical, using digital processing algorithms. A combination of such algorithm, was used to synchronize the speech of several talkers, each of whom had independently recorded the same sentence; the result is choral speech. To align the speech of talker A to the timing of talker B: (a) Calculate a log LPC spectrum each centisecond for each sentence; (b) correlate each of A's spectra with each of B's spectra; (c) find a dynamic time warp "best path" through the matrix of correlations; (d) estimate pitch for talker A every centisecond; (e) speed and slow A's speech according to the time warp found in (c), using a pitch-dependent algorithm to maintain naturalness. Algorithms used will be described and/or referenced. Sentences recorded before and after warping will be played to demonstrate their naturalness, and choral speech will be played to demonstrate the effectiveness of the synchronization.

Break

3:18

TT9. GEST: A computational model for speech production using dynamically defined articulatory gestures. Catherine P. Browman, Louise Goldstein, Elliot Saltzman, and Caroline Smith (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

A preliminary model of speech production based on the concept of a gestural score, that is, an organization of dynamically defined articulatory gestures, has been developed. These gestures, which also serve as linguistic primitives, are coordinated movements of the lips and jaw or tongue and jaw. The movements are generated using a set of underlying dynamic parameters that are specified for each gesture in the score, namely, stiffness, rest position, and damping. (We assume a critically damped linear second-order system with unit mass.) [Work supported by NSF.]

3:30

TT10. Temporal organization of articulatory movements—A multidimensional complex spring model for running speech. O. Fujimura (AT&T Bell Laboratories, Murray Hill, NJ 07974)

A general descriptive framework of speech timing is proposed using a hierarchical linear (elastic) representation of speech units, with a loose linkage between different dimensions (articulators). Inherent temporal properties of phonetic units are represented by their spring constants, and prosodic properties of utterance units are represented by additional springs associated with relevant substructures. For describing a process of speech utterance, an open-ended sructure can be defined by terminal forces in place of position specifications, continuously building up the utterance structure as the speaker produces (or preprograms) the speech signal. Some examples of data from microbeam data will be discussed from this point of view, using "iceberg" patterns as timing marks [O. Fujimura, "A Linear Model of Speech Timing," in *Ilse Lehiste Festschrift*, edited by L. Shockey and R. Channon (Foris Publications, to be published)].

3:42

TT11. Modal analysis of the vocal tract. Ian M. Firth (Department of Physics, University of St. Andrews, St. Andrews, Fife KY16 9SS, Scotland)

The modes of oscillation of the vocal tract (the formants) have been investigated for the first time using the method of modal analysis. While this technique is generally used to study the modal properties of mechanical structures, preliminary studies have shown that this powerful method can be applied to the structure of the air column of the vocal tract. The analysis was carried out by applying an impulse above the larynx on the outside of the throat with closed glottis, and measuring the transfer response at an array of points inside the mouth. Modal analysis presents speech modes in animated form, which allows a comprehensive understanding of their detail to be obtained which has not been available in speech research before. For one subject, details of the modes of the first eight formants to 6.4 kHz will be presented. The first five formants are standing waves propagating in one dimension along the tract, but the sixth to the eighth formants show distinct propagation in two dimensions. Formant frequencies and bandwidths are available from the modal properties. The method has the capability of providing data on the three-dimensional vibration of the tract which has not been available before. The application of this technique to other associated investigations in speech will be mentioned.

3:54

TT12. Detailed spectral analysis of a female voice. Dennis H. Klatt (Room 36-523, Massachusetts Institute of Technology, Cambridge, MA 02139)

Several thousand DFT magnitude spectra have been produced for a selected sample of speech from a single female speaker having a pleasant voice quality. The speaker sustained a number of different vowels while undertaking different laryngeal gestures including (1) slow 1-oct pitch glides up or down and (2) reiterant imitations of different sentences, using either [?V] of [hV] replacements for the pattern of stressed and unstressed syllables in each sentence, where [V] is one of six English vowels. An attempt has been made to quantify the effects on the harmonic spectrum of changes to fundamental frequency, syllable stress, and position of the syllable in the utterance. Also of interest is the detailed way that voicing is initiated and terminated in laryngealized versus breathy onsets and offsets. Previous analyses that used inverse filtering to study the glottal waveform seem to have missed several important aspects of the source spectrum and its change over time. Our analysis reveals the presence of considerable random breathiness noise at frequencies above 2 kHz over portions of many utterances. The strength of the fundamental component and of the various formant peaks relative to overall rms energy in the signal under all of the conditions outlined above have been quantified. There is variation in both the general tilt of the harmonic spectrum and the strength of the fundamental component depending largely on the (presumed) degree to which the larynx is spread/constricted. Locations of spectral zeros have been studied and related to the duration of the open part of glottal period. Synthesis has also been attempted using a modified version of the Klattalk synthesizer, which provides direct control over voicing source open period, source spectral tilt, breathiness noise, and degree of instability in the fundamental frequency contour (the latter parameter being particularly important for natural synthesis of a vowel sustained at constant pitch). Sentences spoken by replacing all of the syllables by [?V] were somewhat easier to synthesize—than sentences spoken by replacing all of the syllables by [hV]. The reason appears to be that in the latter materials, additional spectral peaks—presumably related to tracheal resonances—were sometimes observed in the vowel spectra. Except for this possibly important aspect of voice quality, the synthesis results are very encouraging; source control parameters and formant parameters are monotonic slowly varying continuous functions, suggesting that rules for the synthesis of more natural female voices might be formulated as a next step. [Work supported in part by NIH.]

4:06

TT13. Aeroacoustic considerations for phonation. R. S. McGowan (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The source of sound called "voice" should be characterized from a fluid mechanical point of view. The aeroacoustic theories developed for engineering noise problems can provide the connection between the fluid mechanics and the acoustics of the voice source. The high-impedance characteristics of the piston, which is usually pictured as the source, appears correct. However, the piston velocity should not be equated to the fluid particle velocity at the glottis, since the fluid particle velocity at the glottis is not the acoustic particle velocity. Rather, the oscillating jet at the glottis serves to create pressure fluctuations necessary for acoustic propagation. As a result, only a fraction of the fluid energy is converted into acoustic energy at the glottis. A few basics for this point of view will be presented, while a detailed model awaits development. [Work supported by NINCDS.]

4:18

TT14. Oscillation threshold conditions in phonation. Ingo R. Titze (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242 and the Recording and Research Center, Denver Center for the Performing Arts, 1245 Champa Street, Denver, CO 80204)

A small-oscillation theory is developed for determining threshold conditions for onset and release of vocal fold vibration as a function of lung pressure, tissue viscosity, glottal aperture, and surface wave velocity in the mucosa (vocal fold cover). It is found that *oscillation threshold lung pressure* increases in proportion to mucosal wave velocity. In other words, phonation initiates at lower pressures (less effort) if the vocal fold cover is lax and propagates slow surface waves. Experimental evidence on excised dog larynges confirms that a stiffer cover (at greater vocal fold lengths) requires greater lung pressure. The theoretical prediction that increased glottal aperture (spreading of the vocal folds, as in breathy voice) also increases the oscillation threshold lung pressure has been more difficult to verify experimentally, but increased tissue viscosity produces an observable increase. Applications to phonetics and voice care are cited. [Work supported by NINCDS.]

4:30

TT15. A truncation model for the *F* 2 trajectory in syllables with complex vocalic components. Hongmo Ren (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

Phonetic orthodoxy treats the acoustic realization of a CV_n syllable (N.B. V = vowel or glide) as the concatenation of consecutive transitions from one element to the following one in the syllable. This study proposes an alternative view of the acoustic structure of this type of syllable, based on LPC trackings of $F2$ in sets of selected Chinese and English CV_n syllables. The basic hypothesis is that the underlying $F2$ transitions between any two phonologically adjacent elements in the same syllable (e.g., the $C \rightarrow V_1, V_1 \rightarrow V_2$,...) all originate at the same temporal position—the syllable initiation. Each transition has a specified rate. The acoustic realization will be a programmed "truncation" process; that is, the phonologically preceding transition truncates the phonologically following transition (e.g., the $C \rightarrow V_1$ transition truncates the $V_1 \rightarrow V_2$ transition; the truncated $V_1 \rightarrow V_2$ in turn truncates the $V_2 \rightarrow V_3$ transition). This model provides elegant accounts and quantitative predictions of the undershooting and overshooting phenomena in acoustic target realization. [Work supported by NSF.]

4:42

TT16. A dynamic analysis of reiterant speech production in three languages. Eric Vatikiotis-Bateson (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510) and J. A. S. Kelso (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510 and Center for Complex Systems, Florida Atlantic University, Boca Raton, FL 33431)

Previously, supralaryngeal articulatory motion in reiterant speech produced by speakers of English and Japanese in terms of an underlying dynamical model [Kelso *et al.*, J. Acoust. Soc. Am. **77**, 266–280 (1985); Vatikiotis-Bateson and Kelso, J. Acoust. Soc. Am. Suppl. 1 **78**, S38 (1985)] has been analyzed. Here, the analysis is extended to French, a language whose temporal structure is believed to be different from that of English or Japanese, and a preliminary comparison of movements of the lower-lip/jaw complex for speakers of the three languages is presented. The cross-language comparison is based on articulatory movement data from 13 speakers (five English, five Japanese, and three French). Analysis of reiterant /ba/ and /ma/ productions reveals that French speakers make language-specific linguistic (e.g., stress) and performance (e.g., speaking rate) distinctions that are qualitatively similar in terms of kinematic relations to those observed for Japanese and English. Quantitative differences due to the intrinsic temporal properties of each language are also evident (e.g., in the slope of the peak velocity-displacement relation). It is suggested that the same abstract dynamical model, appropriately parameterized for each language, can characterize the observed articulatory behavior. [Work supported by NINCDS and ONR.]

4:54

TT17. Testing parameters of vocal tract shape. Peter Ladefoged (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

In a mini-version of a system for modeling the speech production process on a Macintosh computer, new algorithms have been devised for interpreting values of eleven control parameters: front–back tongue movement; high–low tongue movement; jaw opening; tip of tongue raising; tip of tongue advancing; lip height; lip rounding and protrusion; velic opening; advancing of the tongue root; rhotacization; and larynx raising. New algorithms have also been devised for converting the vocal tract shapes into sets of tubes, whose lengths and areas can be used as a basis for calculating the sounds that would be produced. The phonological features nasal and advanced tongue root can be given direct values in this set of commands. The features high–low and front–back can be interpreted as auditory specifications from which corresponding tongue shapes can be calculated. It is not known if it is possible to devise algorithms for interpreting other features in terms of these commands.

THURSDAY AFTERNOON, 11 DECEMBER 1986 — SALONS G AND H, 1:30 TO 5:00 P.M.

Session UU. Underwater Acoustics VIII: Remote Sensing and Underwater Acoustics, Part 2

Stephen Piacsek, Chairman
SACLANT ASW Center, 19026 La Spezia, Italy

Contributed Papers

1:30

UU1. Remote sensing of the upper ocean with sound: A design study. Nicholas P. Chotiros (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

In order to obtain a better understanding of ocean dynamics, it is desirable to be able to observe the mechanical coupling between the kinetic processes of the ocean, such as surface motion, turbulence, and internal waves. The important issues in designing a sonar for this purpose are presented. The objective is not only to measure the statistics of the processes, such as mean velocity or rate of kinetic energy dissipation, but also to track the movement of discrete water masses in three dimensions. The critical design issues and signal processing requirements will be discussed. [Work supported by ONR.]

1:45

UU2. The effect of thermohaline steps on range-dependent acoustic propagation near the northeast coast of South America using high resolution data. Stanley A. Chin-Bing, David B. King, and Janice

D. Boyd (Naval Ocean Research and Development Activity, NSTL Station, MS 39529-5004)

The effects of large thermohaline steps on acoustic transmission loss (TL) for a range-independent sound velocity profile (SVP) near the northeast coast of South America were reported [Chin-Bing and King, J. Acoust. Soc. Am. Suppl. 1 **76**, S84 (1984)] using a high resolution SVP (in depth), a discrete standard depth SVP, and a depth thinned SVP. The thermohaline step structure created adjacent regions of near isovelocity connected by a high gradient small transition region; this "step" structured SVP affected the TL even at low frequencies. The study is now extended to include recent experimental high resolution range-dependent thermohaline data. Range-dependent TL comparisons are made using SVPs obtained from (1) the high resolution data, (2) an oceanographic forecast model, (3) discrete standard depths, and (4) a depth thinned profile. This range-dependent study supports the basic finding from the range-independent study—TL structure can be adversely affected by failing to include the detailed SVP step structure. Furthermore, oceanographic forecasts models that provide SVPs need a high degree of accuracy in regions where these vertical anomalies exist.

2:00

UU3. Waveguide characterizing and source localization in a shallow water waveguide using Prony's method. E. C. Shang,[a] Z. Y. Haung, and H. P. Wang (Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China)

Horizontal samplings of a point source in a shallow water waveguide are processed by Prony's method. First, when the source position is known, a high resolution estimation of the normal mode parameter k_m—the horizontal component of mode wavenumber, can be obtained. Owing to the weak damping process, β_m, the modal attenuation coefficient can be obtained by using a modified Prony's method. In this paper, an "interval average and iterative" approach is proposed. A numerical example illustrates that, a 10^{-4} resolution of km (150-Hz frequency) is provided by a 500-m data length, roughly an order of magnitude smaller than the data length required by the Hankel transformation technique. Second, when the source position is unknown but the waveguide characters (k_m, β_m) are known, the same procedure can be used for source position estimation. Some numerical examples in typical shallow water waveguide will be presented. [a] Current address, Department of Computer Science, Yale University, New Haven, CT 06520-2158.

2:15

UU4. Esimation of seafloor geoacoustic parameters from cross correlation of measured and simulated broadband signals. Paul J. Vidmar and David W. Oakley (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

The ratio R of the compressional (P) wave velocity to water sound speed, and g, the gradient of P wave velocity, have been measured for an abyssal plains environment by maximizing the similarity between measured and simulated low-frequency broadband acoustic signals from a single shot (explosive source) at long range. Simulated time series were produced by a computational model that predicts low-frequency bottom interacting waveforms that agree well with small grazing angle acoustic data from this environment [D. P. Knobles and P. J. Vidmar, J. Acoust. Soc. Am. **79**, 1760–1766 (1986)]. Our measure similarity is C_{max}, the maximum of the absolute value of the cross-correlation function of two time series, and C_{max} has a strong dependence on R, g, and the experimental geometry; it has a weak dependence on the sediment density and P wave attenuation. Estimates of R and g agree with reported values. [Work supported by the NORDA Bottom Interaction Program.]

2:30

UU5. The frequency and depth dependence of geoacoustic parameters in a stratified sea bottom, remote-sensed from shallow water acoustic propagation. Ji-xun Zhou,[a] Peter H. Rogers, and Jacek Jarzynski (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Due to the difficulty of direct measurement, inverse techniques need to be developed for determining the bottom geoacoustic parameters in low- and middle-frequency ranges. In this paper, dispersion analysis and normal mode measurements in shallow water are extended to extract acoustic attenuation and speed in a horizontally stratified bottom versus frequency and depth. The uncertainty is discussed. The computational and experimental results show that for a given frequency band, we can find an equivalent depth profile of sea-bottom acoustic attenuation with a linear frequency dependence (without depth structure) on some field characteristics, such as the attenuation rate of individual mode, the frequency response of long-range sound propagation, and the amplitude ratio of mode 2 to mode 1, etc. But the resultant equivalent negative gradient for sea-bottom attenuation is too strong to be accepted in comparison with previously available data. The conclusion is that nonlinear frequency dependence of the acoustic attenuation in the upper sedimentary layer is required to explain many aspects of shallow water sound propagation. [a] On leave (1985) from Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China.

2:45

UU6. Waveform inversion to extract permeability and porosity of marine sediments. Altan Turgut and Tokuo Yamamoto (Division of Applied Marine Physics, Rosenstiel School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, FL 33149-1098)

Propagation of acoustic pulse through marine sediments is modeled by the Biot theory. It is shown that the dispersion and attenuation of acoustic waves through a marine sediment are unique functions of porosity and permeability. A waveform inversion scheme has been developed to extract permeability and porosity from pulse deformation measurements. Uniqueness and consistency of the inversion have been examined using synthetic data. It is concluded that the proposed inverse method is useful for remote sensing of permeability and porosity of marine sediments. [Work supported by ONR.]

3:00

UU7. Results from remote sensing experiments of bottom shear modulus profile using a passive BSMP system. Tokuo Yamamoto, Craig Conner, Mohsen Badiey, and Altan Turgut (Division of Applied Marine Physics, Rosenstiel School of Atmospheric Science, University of Miami, Miami, FL 33149-1098)

Recently, an entirely new method of a passive remote system BSMP to extract the bottom shear modulus profile from the measured bottom motion induced by surface gravity waves [T. Yamamoto and T. Torii, Geophys. J. R. Astron. Soc. **85**, 413–432 (1986)] was developed. The BSMP inversion has been proven to be unique and consistent. Pilot systems of BSMP have been built and tested at the Great Bahama Bank, the Ohi River Delta, and the New Jersey Shelf. The shear modulus profiles of a wide range of bottom sediments and rocks and of complex layered structures have been extracted by the BSMP systems and have been supported by direct measurements. In this paper, experimental procedures and results from these BSMP experiments are discussed. [Work supported by ONR.]

3:15

UU8. Angular dependence of acoustic backscattering over a Mn nodules field. D. Alexandrou and C. de Moustier (Scripps Institution of Oceanography, A005, La Jolla, CA 92093)

The in-phase (I) and quadrature (Q) components of the seafloor backscattered acoustic returns received by a Sea Beam multibeam echosounder over a manganese nodules field in the tropical northeastern Pacific have been digitized and recorded on magnetic tape. Because both the I and Q components of the signals were present, the sidelobe interference inherent in multibeam systems [C. de Moustier, J. Acoust. Soc. Am. **79**,

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316–331 (1986)] was able to be removed, and a complete angular dependence function of acoustic backscattering between $\pm 20°$ of vertical incidence was obtained. These results and the sidelobe interference removal method are discussed. [Research funded by ONR.]

3:30

UU9. The Biot–Stoll model: An experimental assessment. Burlie A. Brunson and Charles W. Holland (Planning Systems, Inc., 7900 Westpark Drive, Suite 600, McLean, VA 22102)

A promising approach in the prediction of acoustic wave speeds and attenuations in marine sediments is the use of the Biot theory as implemented by Stoll. The accuracy of the Biot–Stoll sediment model was examined for a variety of natural marine sediments taken from four shallow water sites in the Mediterranean Sea, including silty–clay, sand, and gravel. Comparisons were made between the Biot–Stoll model predictions of compressional velocity, compressional attenuation, and shear velocity with *in situ* and laboratory measurements. The model predictions on the whole showed excellent agreement with the measured data.

3:45

UU10. Ray-parameter method for acoustic velocity measurements of sediment layers. Shuying Zhang (Shanghai Acoustics Laboratory, Academia Sinica, 456 Xiao-Mu-Qiao Road, Shanghai, People's Republic of China)

The ray-parameter method for acoustic velocity measurements of sediment layers presented by Bryan [*Physics of Sound in Marine Sediments*, edited by L. Hampton (Plenum, New York, 1974), pp. 119–130] has been extended to the case of layers having a common dip angle ϕ. The expressions for calculating the velocity V_2 and thickness Δz of the second layer are

$$V_2 = \left(\frac{(x_2 - x_1)\cos[\sin^{-1}(\lambda V_1)]}{\lambda(t_2 - t_1)\cos[\sin^{-1}(\lambda V_1) - \phi] - \lambda(x_2 - x_1)\sin\phi/V_1} \right)^{1/2}, \quad (1)$$

$$\Delta Z = \frac{V_2}{2}(1 - \lambda^2 V_2^2)^{1/2}\left((t_2 - t_1) - \frac{(x_2 - x_1)\sin\phi}{V_1\cos[\sin(\lambda V_1) - \phi]}\right), \quad (2)$$

where the velocity of the first layer (V_1) is premeasured, and the parameters λ, $(x_2 - x_1)$, and $(t_2 - t_1)$ are determined by a certain data processing procedure from seismic records. The errors of measuring velocities caused by uncertainties of λ and ϕ can be estimated as:

$$\left(\frac{\delta V_2}{V_2}\right)_\lambda = \frac{1}{2}\frac{\delta\lambda}{\lambda}\left(\frac{Z_1}{\Delta Z}\frac{\tan\theta_1}{\tan\theta_2}\cos^2\theta_2 - \sin^2\theta_2\right), \quad (3)$$

where $\theta_2 = \sin^{-1}[(V_2/V_1)\sin\theta_1]$, θ_1 is the angle of incidence to the layer, and z_1 is the altitude of the receiving streamer above the layer;

$$\left(\frac{\delta V_2}{V_2}\right)_\phi = \frac{\delta\phi}{2}\,\frac{V_1(t_2 - t_1)\sin(\theta_1 - \phi) - (x_2 - x_1)\cos\phi}{V_1(t_2 - t_1)\cos(\theta_1 - \phi) - (x_2 - x_1)\sin\phi}. \quad (4)$$

4:00

UU11. The determination of layer thickness in layered ocean bottoms using acoustic fields. Michael F. Werby and Hassan B. Ali (Naval Ocean Research and Development Activity, NSTL, MS 39529)

For a shallow water environment, it is theoretically possible to extract information on properties of the ocean bottom from the beamformed response of a cw source. In particular, using synthetic aperture offset measurements from a cw source (for discrete frequencies in the range 20–100 Hz), one can obtain information on bottom layering and the associated compressional and shear velocities. The technique rests on the correspondence between the maximum beamformer response in angular space and the normal mode response in wavenumber space. The present work extends the preceding method by using the properties so determined to deduce the thickness of the relevant layers. The technique is based on the comparison of results of calculations for the unknown layer with those obtained from a "reference" layer. Examples of the method are presented.

4:15

UU12. Reflection tomography imaging of submerged objects. Charles F. Gaumond and Phillip B. Abraham (Naval Research Laboratory, Code 5132, Washington, DC 20375-5000)

In a continuing study of acoustic reflection tomography [C. F. Gaumond and P. B. Abraham, J. Acous. Soc. Am. Suppl. 1 **78**, S80 (1985)], results obtained from synthetic data for prolate spheroids will be presented. These will be compared with similar results for various spherical scatterers. This novel method of image reconstruction of strong scatterers will be briefly reviewed.

4:30

UU13. An ultrasonic ranging system for robot position sensing. Fernando Figueroa and John S. Lamancusa (Department of Mechanical Engineering, The Pennsylvania State University, University Park, PA 16802)

Currently used robot position control systems consider only the joint angles when positioning an end effector in space. This approach can lead to serious errors due to structural deflection under load, backlash in joints, manufacturing tolerances, etc. A method is needed to directly measure the absolute position and orientation of the end effector in three-dimensional space in order to achieve positioning accuracy. An ultrasonic ranging system is proposed to accomplish this task. It consists of: (1) one or more transmitters mounted on the robot's end effector; (2) receivers (four or more required), mounted adjacent to the robot's work envelope; and (3) analog and digital electronics to drive the transducers, process the received signals, and calculate the position of the end effector. The end effector's position is calculated using the time of flight measured from the emitter to each receiver, by an intersection of spheres algorithm. The performance objectives of this system are: (a) measure position with accuracy of ± 0.001 in. (b) up to a range of 3 m. and (c) with at least 1-Hz update rate. A theoretical error analysis of the system is presented. Types of emitter signals are compared, operating frequency considerations are discussed, and hardware recommendations are made. [Work supported by NSF.]

4:45

UU14. Bivariate swimbladder size allometry properties of mesopelagic fish and bioacoustic prediction. R. Alfred Saenger (Naval Underwater Systems Center, New London, CT 06320)

One of the fundamental problems in modeling the low-frequency contributions to scattering strength of swimbladder fish is to transform a measured or theoretical size-depth distribution $h(l\,|z)$ for an individual species to the corresponding distribution $q(R\,|z)$ of equivalent swimbladder radius R. This is usually done using a y-on-x regression line relation for the species, where $y = \ln R$, $x = \ln l$, and l = fish standard length. Analysis of archival Ocean Acre swimbladder size allometry data for 38 species indicates that this procedure is physically inadequate. Due to biological variability, a wide range of R values is associated with a fish of length l. Since resonance scattering from gas-filled swimbladders is depth dependent, important scattering contributions may come from fish with bladder sizes R not lying on the regression line. To avoid systematic prediction errors, the x,y distribution $p(x,y)$ rather than the regression line must be used to calculate $q(R\,|z)$. This transformation problem and its modeling consequences are examined in light of the fact that $p(x,y)$ for many swimbladder species has been found to be bivariate normal.

Session VV. Musical Acoustics VII: Ethnic Musical Instruments, Part 2: Percussion and Winds

Edith L. R. Corliss, Chairman
Forest Hills Laboratory, 2955 Albemarle Street, N.W., Washington, DC 20008

Invited Papers

9:00

VV1. Acoustics of Oriental gongs and bells. Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Many different types of bells and gongs are found throughout the Far East. Each has an interesting story to tell the acoustician as well as the musicologist. Among the instruments we will discuss are gamelan gongs from Indonesia, opera gongs from China, gongs from Thailand, temple bells from Japan, and chime bells from China.

9:30

VV2. Twelve Nasca panpipes. Joerg Haeberli (38 Sylvan Drive, Morris Plains, NJ 07950)

Tonometric meaurements for 12 Nasca panpipes, two of which can be dated, are presented. The acoustical properties of the two types of identified pipes differ significantly. Specific frequencies reoccur among different instruments. This allowed building of a sequence of steps which are based on equal frequency intervals in hertz, rather than on a logarithmic series. The accuracy of tuning was better than 10 Hz for over 60% of the pipes. This is adequate for panpipes that are activated by mouth blowing.

10:00

VV3. The ancient Chinese "calendrical" pitchpipes. Ernest G. McClain (P.O. Box 127, Belmont, VT 05730)

The most important symbol of "order" or "natural law" in Han China (202 B.C. to 220 A.D.) was a set of 12 end-blown bamboo tubes known today as the "pitchpipes." Each pipe sounded a single tone of the chromatic octave and served as "tonic" for a specific calendrical period. All weights and measures of the empire were derived from the dimensions and capacity of the fundamental pipe, from which the lengths of other pipes were proportioned *according to string ratios.* It is of interest to the history of acoustical physics and to the study of science and civilization in China to ask how much "order" such pipes display when blown. Within a narrow Han limitation to diameter/length ratios between 1/30 and 1/27, and to fundamental lengths somewhat shorter than the foot, the cyclic excess in "pure" wholetones of 9:8 offsets pipe "end correction" sufficiently to permit the wholetone scale on the tonic (i.e., the six *yang*, "male" tones, the only ones strictly designated *lu*, meaning "order") to be sounded at or very near equal temperament values. Thus pitchpipes—within well-defined physical parameters—were arguably an appropriate cultural icon for *cyclic orderliness.* Sinologists overlook this lawfulness.

Contributed Papers

10:30

VV4. Primitive musical instruments using Helmholtz resonators. Ian M. Firth (Department of Physics, University of St. Andrews, St. Andrews, Fife, KY16 9SS, Scotland)

Measurements will be presented on some primitive musical instruments that produce sound from excitations of a Helmholtz resonator. Input acoustical response measurements will be compared with the sound output when the instruments are blown. In some instruments, pitch can be altered by changing finger position, but others have constant pitch. The complicated pigeon whistle which is attached to the legs of pigeons will be described. This lightweight whistle contains a large Helmholtz resonator and several quarter wave pipes, and is excited in flight by air streaming across sharp lips on each of the parts.

10:45

VV5. Characteristics of shakuhachi signals. Linda A. Seltzer (Department of Music, University of California, Berkeley, CA 94709 and Recording and Research Center, Denver Center for Performing Arts, Denver, CO 80203)

Selected portions of a recorded performance of shakuhachi (Japanese bamboo flute) music have been analyzed using spectrograms, and waveforms have been plotted by computer. The study has revealed interesting and unusual features of the signal during special performance practices. These effects add variety to the timbre, as well as subtlety in the conception of pitch in the music. The effects include: nonharmonic partials which are anticipations of harmonics of the subsequent note; harmonics which linger and become nonharmonic partials of the subsequent note;

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90-deg phase shifts when a lateral opening is repeatedly hit with a finger; subharmonics during transitions between notes; gradual, smooth microtonal variations of pitch in the course of long notes; and abrupt changes in timbre.

11:00

VV6. Vibrational modes of a Chinese two-tone bell. H. John Sathoff (Bradley University, Peoria, IL 61606), B. R. Richardson (University College, Cardiff, Wales), D. Scott Hampton, Thomas D. Rossing (Northern Illinois University, DeKalb, IL 60115), and André Lehr (Royal Eijsbouts Bellfoundry, Asten, The Netherlands)

Many ancient Chinese bells, by virtue of their oval shape, sound two distinctly different tones, depending upon where they are struck. These so-called *sui* and *gu* tones usually differ in pitch by about a minor third. The vibrational modes and sound radiation from a carefully dimensioned modern copy of a Chou dynasty (1030–221 B.C.) bell in the Dutch National Carillon Museum have been studied. The lowest modes, and many of the higher ones as well, occur in pairs, with one member having an antinode near the *sui* strike point and the other near the *gu* strike point. The frequency separation of these doublets is determined by the shape of the bell. The modal shapes, determined by holographic interferometry, are similar to those observed in round church bells, but with more mixed modes due to the greater damping.

11:15

VV7. Acoustics of Caribbean steel drums. Thomas D. Rossing, D. Scott Hampton, and Joshua Boverman (Northern Illinois University, DeKalb, IL 60115)

As part of an ongoing study of Caribbean steel drums, the acoustical behavior of three alto/double tenor steel drums, each having 14–17 notes in the range of $F_3^\sharp$ to C_6 (185–1047 Hz) has been studied. All except for the highest notes have three prominent harmonic partials, each associated with a mode of vibration of the local area of the drum tuned to that note. Additional harmonic partials (up to eight) are identified when a note is struck strongly, suggesting a nonlinear behavior characteristic of shallow spherical-cap shells [N. H. Fletcher, J. Acoust. Soc. Am. **78**, 2069–2073 (1985)]. By means of holographic interferometry, some modes of vibration have been able to be identified as "local" and some as "global," encompassing several note areas on the drum.

11:30

VV8. Tonal design and tuning of Caribbean steel drums. Clifford Alexis, Al O'Connor, and Thomas D. Rossing (Northern Illinois University, DeKalb, IL 60115)

Steel drums were invented in Trinidad in the 1940s, but they soon became popular throughout the Caribbean islands. The drums in a steel band are usually fabricated from 55-gal oil drums, and they are known by various names, such as soprano, ping pong, alto, double tenor, guitar, cello, and bass. Each drum may have from 4–32 different notes. A skilled steel drum maker tunes at least one overtone of each note to a harmonic of the fundamental. Whenever possible, a second harmonic is tuned as well.

FRIDAY MORNING, 12 DECEMBER 1986 SALONS 2 AND 3, 9:00 TO 11:35 A.M.

Session WW. Noise VIII: Sound Intensity and Sound Power, Part 1

Richard J. Peppin, Chairman
Scantek, Inc., 1559 Rockville Pike, Rockville, Maryland 20852

Joseph Pope, Co-Chairman
Bruel and Kjaer, Inc., 185 Forest Street, Marlborough, Massachusetts 01752

Chairman's Introduction—9:00

Invited Papers

9:05

WW1. Precision of sound power measurements with intensity technique. Jiri Tichy (Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802)

The precision of the sound power determination by measuring acoustic intensity depends on the precision of the measurement system, selection of the measurement area, and the number of measurement points. The decision on the optimum choice of these parameters depends on the properties of the radiated sound field and on the environmental effects including simultaneous sound radiation from other sources. To obtain qualifying information on the sound field properties, the measurement of the complex acoustic intensity may be necessary. To quantify the judgment on the measurement parameter selection, a series of indicators has been recently suggested. This presents an analysis of the mutual effects of the measurement parameters and their effect on the precision of the sound power determination.

9:35

WW2. Calibration of microphones for sound intensity measurements. A. F. Seybert (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506-0046)

The two-microphone sound intensity method is potentially a valuable tool for noise source identification, sound power determination, and for other measurements (e.g., transmission loss) that are based on sound intensity measurements. An essential ingredient in an accurate sound intensity measurement is the accurate measurement of the phase between the two microphones. Because the phase is generally small, on the order of a few degrees, the microphones must either be "phase matched," or a phase calibration procedure must be used. This paper outlines the design and test procedures for using a plane-wave tube to perform a microphone phase calibration with an uncertainty of less than approximately 0.2 deg. The phase calibration procedure is illustrated for a pair of condenser microphones and for a pair of miniature crystal microphones typically used in hearing aids.

10:05

WW3. Instrumentation for intensity measurements. Torben G. Nielsen (Brüel & Kjaer, DK-2850 Naerum, Denmark)

The concept of sound intensity has been known for many years, and the usefulness of an intensity measuring device has been understood for just as long a time. The factor that has drawn the attention of the noise control community towards intensity measurements within the last decade is the development of instruments making direct measurement of sound intensity a feasible task in the field as well as in the laboratory. Today, a selection of analyzers and probes of different types are being used for practical intensity measurements. The analyzers fall mainly into three groups: dual-channel FFT analyzers, real-time analyzers based on digital filters, and small battery-operated systems usually equipped with analog filters. Most probes are based on the two-microphone principle, but a system using a pressure and a velocity transducer is also available. From a user's point of view, these different types of instruments should provide identical measurement results—but this is not always the case in practice. In the presentation, the various instrument principles are briefly reviewed and the performance of the instruments when subjected to different sound fields and signal types is discussed.

Contributed Papers

10:35

WW4. Errors introduced by intensity probes. Per V. Brüel (Brüel & Kjaer, 18 Naerum Hovedgade, DK-2850 Naerum, Denmark)

The paper describes the errors introduced in intensity measurements on account of the finite size of the probe. The reflections between the microphones cause errors not only in the pressure amplitude, but also give rise to uncontrollable phase changes which result in significant errors. This is illustrated by systematic measurements on some of the most commonly used two-microphone probes, and measures to be taken for minimizing these errors are indicated. Furthermore, the types of errors caused when the points of measurement of the sound pressure and the particle velocity are not coincident, are also shown.

10:50

WW5. Instantaneous intensity. Richard C. Heyser (10415 Fairgrove Avenue, Tujunga, CA 91042)

Expressing time as a fourth space coordinate under the rule $x_4 = ict$, where i is the imaginary operator, allows the interpretation of a complex velocity which is the four-dimensional gradient of a potential. The three-conventional space components of this gradient yield particle velocity, while the fourth component yields sound pressure divided by the product of density and sound speed c. In accordance with an energy theorem of Heyser [J. Audio Eng. Soc. **19**, 902 (1971)], this four-dimensional velocity is interpreted as the local component of an energy function whose global support is represented by a complex Hilbert transform component. This leads to a total energy density which contains a term identifiable as the reactive component of an instantaneous complex intensity. Coinciding with the well-known expression for the time averaged intensity of monochromatic signal dependence, this instantaneous intensity applies to any time dependence and has a time average identical to the conventional expression. It is shown that, in the case where instantaneous reactive intensity vanishes, the instantaneous active intensity is numerically identical to the energy–time curve (ETC) which is now used in acoustical analysis.

11:05

WW6. One-microphone sound intensity method. Anna L. Mielnicka-Pate and Robin M. Bai (Department of Engineering Science and Mechanics, Iowa State University, Ames, IA 50011)

A new method based on the cubic spline approximations of pressure gradient was developed for particle velocity and acoustic intensity calculations. In this method, a cubic spline is interpolated for every four consecutive pressure measurements performed with equal spacing. In contrast to conventional approximate methods, only one microphone and synchronized scanning are required in this technique. A numerical simulation was performed in order to evaluate the accuracy of this method. A monopole and a baffled piston were chosen as the acoustic sources in this simulation since the analytical solutions are easily available for comparison purposes. The cubic spline interpolation technique provided substantially better accuracy than the finite difference method. In addition, some preliminary experimental data will be presented for the cubic spline and finite difference techniques. [Work supported by NSF.]

11:20

WW7. Complex wave fronts and generalized phase speed described by the active intensity vector. Adin Mann and Jiri Tichy (Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802)

The active intensity vector is directed normal to the surfaces of constant phase in an acoustic field. The surfaces of constant phase are wave fronts; consequently, the active intensity vector characterizes the resultant wave propagation at a point in space. This physical description provides a powerful means to organize a complicated nearfield of a radiator or a scatterer. In the farfield, the phase speed of the resultant wave fronts is the plane-wave phase speed c. However, in general, the phase speed varies throughout an acoustic field. The general phase speed C_p is described by taking the total derivative of the phase of the pressure. The resultant wave fronts and their phase speed will be discussed with examples of acoustic vorticies, and the measured fields diffracted by a rigid circular disk or generated by a loudspeaker.

Session XX. Physical Acoustics VII: Theoretical and Computational Problems

Mark F. Hamilton, Chairman
Department of Mechanical Engineering, University of Texas, Austin, Texas 78712-1063

Contributed Papers

8:30

XX1. On the adequacy of linear acoustics for prediction of acoustic pressure transients created by laser beams moving over water surfaces at sonic speeds. Allan D. Pierce and Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Previous discussions of laser-generated sound, particularly with reference to themoacoustic effects, are often based on linear equations. However, seemingly plausible arguments can be developed to the effect that the peak acoustic pressure predicted by a linear theory will grow without bound when the sound is generated by a laser beam that travels for an indefinitely long period of time over a water surface at exactly the speed of sound. One consequently asks (1) whether the linear model does indeed predict a pressure singularity for a sonically moving laser beam, and (2) whether nonlinear effects can be neglected. Analysis by two different methods (Fourier transforms and method of retarded potentials using the transient Green's function) shows that the linear acoustic framework does not predict any singularity in the acoustic pressure generated by a thermoacoustic source moving at the speed of sound over a water surface. Additional theoretical considerations strongly suggest that cumulative nonlinear effects can be ignored during the buildup phase of the acoustic pulse. The key ingredient in the mathematical model, which holds the peak pressure to a finite value, is the pressure release boundary condition at the water surface. However, once the acoustic pulse is launched (i.e., after the laser is turned off), accumulative nonlinear steepening effects may alter the received pressure waveform. In the latter case, a quasi-one-dimensional nonlinear acoustics model based on Burger's equation is expected to be adequate to predict the nonlinear distortion in farfield sound propagation. [Work supported by ONR.]

8:45

XX2. Development of a finite difference model for arbitrary axisymmetric acoustic pulses applicable in investigating finite ground impedance effects. Victor W. Sparrow and Richard Raspet (U. S. Army Construction Engineering Research Laboratory, P. O. Box 4005, Champaign, IL 61820-1305)

With the advent of supercomputers, accurate modeling of sound propagation over a finite impedance ground via finite difference (FD) techniques is becoming possible. In this preliminary investigation, FD solutions to the fundamental hyperbolic partial differential equations governing acoustics which agree with analytic solutions are sought. The particular numerical techniques are similar to those of computational compressible fluid flow, modified for spherical waves in a cylindrical coordinate system to correctly model boundary conditions. Results for acoustic simple sources will be provided.

9:00

XX3. Exterior boundary-value problems for the Helmholtz equation with a generalized impedance boundary condition. Allan G. Dallas (Naval Research Laboratory, Code 5132, Washington, DC 20375-5000)

An exterior boundary-value problem for the Helmholtz equation is considered, the boundary condition being of the (generally nonlocal) form $\partial u/\partial v + Bu = g$, with B a bounded linear operator in the Hilbert space $L_2(\Gamma)$, Γ denoting the boundary of the exterior domain, and g given in $L_2(\Gamma)$. It is also required that B satisfy a certain dissipativeness condition. The basic existence, uniqueness, and continuous-dependence results are stated. Various techniques for construction of the solution are described, which are extensions of the methods known for the Neumann or "classical" impedance boundary-value problems. It is argued that such an exterior problem is the appropriate one to study when modeling the more difficult interior–exterior interface problem corresponding to the scattering of time-harmonic acoustic waves by a penetrable body.

9:15

XX4. Impedance matching in the design of constant directivity horns. James McLean and Elmer L. Hixson (Department of Electrical and Computer Engineering, The University of Texas, Austin, TX 78712)

A model capable of predicting impedance and power transfer characteristics of so-called "constant directivity" (CD) horns was developed and presented at the spring 1986 meeting of the Society [J. Acoust. Soc. Am. Suppl. 1 **79**, S90 (1986)]. It is well known that a CD horn does not exhibit flat frequency response because reflections occurring at the diffraction slot cause resonances in the matching section. This model has been used to apply distributed circuit design techniques to CD horns in order to reduce the resonant peaks in the driving point impedance of a particular CD horn, the JBL 2360. Modifying the matching section of the CD horn yields a more nearly constant and resistive driving point impedance without adversely affecting its constant directivity characteristics. Several alternative designs for the 2360 will be presented and the relative merits of each will be discussed.

9:30

XX5. Numerical solution of Burgers' diffusion equation for arbitrary waveforms: Convolution method. R. H. Mellen (PSI Marine Sciences, New London, CT 06320)

Burgers' nonlinear equation for plane acoustic waves in a dissipative fluid reduces to the diffusion equation by Hopf–Cole transformation. A Bessel series solution has been obtained for an initially sinusoidal wave [D. T. Blackstock, J. Acoust. Soc. Am. **36**, 534–542 (1964)]. Solutions for initial waveforms of arbitrary shape are also possible by numerical methods using the FFT method [R. H. Mellen, J. Acoust. Soc. Am. Suppl. 1 **75**, S32 (1986)]. However, the method is limited to relatively small Goldberg numbers because of the roundoff errors. Gaussian convolution is shown to be an equivalent method that does not suffer this limitation. Examples of calculated waveform distortion for typical cases including random noise are presented.

9:45

XX6. Diffraction of acoustic waves by using the generalized ray theory. C. George Ku (Martin Marietta Baltimore Aerospace, Baltimore, MD 21220)

The diffraction of transient waves by a curvilinear boundary using the theory of generalized rays is developed. The generalized ray integrals, which represent the Fourier transform of diffracted waves, involve a double integration with respect to two wave slownesses. The inverse Fourier transform of these double integrals are computed by applying the simultaneous transformation of variables and the Cagniard method. The phase functions of the integrand provide the ray paths of incident, reflected, and diffracted waves. Since the pulses arrive at a point of observation in successive order, the theory furnishes an exact solution up to the time of arrival of the next ray and enables one to analyze, in detail, the signals recorded by the receiver.

10:00

XX7. Diffraction of a rectangular beam by a crack edge. John G. Harris (216 Talbot Laboratory, 104 S. Wright Street, Urbana, IL 61801)

Ultrasonic transducers, operating at microwave frequencies, radiate highly directional wave fields or beams. These beams are used to probe solids for defects such as cracks. To model this process, the compressional and shear waves reflected from a semi-infinite crack and diffracted from its edge when it is struck by a two-dimensional, compressional beam are calculated asymptotically. A beam whose initial cross section is rectangular is considered. The crack is assumed to lie in the nearfield or in the nearfield–farfield transition region of the incident beam. Thus the incident beam is only slightly divergent when it strikes the crack edge, and it ensonifies only a finite region of the crack surface. But this region acts as if it were a secondary aperture. Therefore, of particular interest in the calculation are the results showing how the reflected beam diverges as it propagates away from this secondary aperture and arrives at the aperture of the incident beam. [Work supported by NSF.]

10:15

XX8. Parametric wave interactions near reflecting surfaces. Corinne M. L. Darvennes and Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

Parametric wave interactions offer the possibility of synthesizing directional transmitting and receiving arrays by exploiting the acoustical nonlinearity of the fluid. Use of parametric arrays is frequently suggested as one way to reduce problems involving multipath in both the transmission and reception of sound in reverberant environments. Novikov *et al.* [Sov. Phys. Acoust. **29**, 138 (1983)] have analyzed the case of parametric radiation parallel to an adjacent reflecting surface. Here their work is reviewed and the effects of reflection on the radiation pattern are discussed. The analysis is then modified to include the case of parametric reception. Of interest is the use of a parametric receiver for calibrating the sound field radiated by surface ships. The proposed application involves mounting the pump on the ship and placing the hydrophone in the farfield. Although the proposed system has recently been analyzed for a free-field environment [M. F. Hamilton *et al.*, J. Acoust. Soc. Am. Suppl. 1 **78**, S40 (1985)], the effects of reflected sound in the interaction region must be understood. [Work supported by ONR.]

10:30

XX9. Reflection from an anechoic duct termination. J. P. Tanzosh (David W. Taylor Naval R&D Center, Bethesda, MD 20084) and S. I. Hayek (The Department of Engineering Science and Mechanics, and the Applied Research Laboratory, The Pennsylvania State University, State College, PA 16802)

A theoretical model was developed which describes the farfield reflection from various absorbent terminations of an acoustic duct with rigid walls. The absorbent termination is utilized to reduce the reflected sound from the ends of ducts and pipes. Such sound absorbing terminations can be used to minimize the shock wave (water hammer) travel in water pipes, reduce the flow noise at flow-through pressure valves and at stations along the hydraulic lines where cross-sectional changes occur. A parametric study of the terminations's geometry and material properties revealed the effects of longitudinal and shear wave resonances inside the termination on the frequency response of the reflection. A partially cemented lower boundary achieves much greater echo reduction than a uniform rigidly attached or stress-free boundary. Peaks in the frequency response result not only from compressional thickness resonances, but also from shear and lateral compressional resonances induced by the discontinuity. A slight impedance mismatch promotes multiple reflections within the termination, hence increased resonance effects. [Work supported by the Naval Sea Systems Command.]

10:45

XX10. A model for transverse wave sensitivity to poor adhesion in adhesively bonded joints. J. L. Rose, A. Pilarski, J. Da-Le (Department of Mechanical Engineering and Mechanics, Drexel University, Philadelphia, PA 19104), and D. LeCuru (Department of Service Controle Non Destructif, Aerospatiale, Suresnes Cedex, France)

Poor surface preparation in adhesive joints of metal-to-metal type usually results in a weak bond. The adhesive strength prediction problem has been studied extensively with little success to date. The authors of this paper are attempting to explain changes in the reflection and transmission coefficients of oblique incident longitudinal and transverse waves accompanying changes in adhesive bond quality, based on a model of a bond with finite rigidity. The reflection transmission characteristics for a solid–solid imperfectly bonded interface were based on an assumption that displacement across the interface was not required to be continuous. This discontinuity was taken to be linearly related to the stresses which are continuous across the interface. A comparison of cases "welded" and "smooth" boundary conditions gave us the optimal angle of incidence of transverse waves with the best sensitivity to areas with poor adhesion. Theoretical and experimental results are compared with excellent agreement.

11:00

XX11. Ray acoustic analysis of plane-wave coupling into large open waveguides: Plane parallel geometry. L. B. Felsen and H. Shirai (Polytechnic University, Route 110, Farmingdale, NY 11735)

When a plane sound wave impinges on a wide open-ended baffled or unbaffled plane parallel waveguide, the behavior of the sound field coupled into the interior differs from that when the waveguide is tubular [H. Shirai and L. B. Felsen, this meeting]. The difference is due to the fact that plane waves retain their integrity between wall reflections. By application of ray theoretic, modal, and hybrid ray-mode methods, one may establish the connection between the geometrically sharply delineated multiple reflected sheet beam, the edge diffracted-multiple reflected ray fields, and the guided modes. For very strongly overmoded conditions, the significantly contributing modes are those whose plane-wave congruences have directions near that of the incident plane wave. Invoking ray-mode equivalences, a hybrid form is found to describe the propagation phenomena in a physically incisive and numerically efficient manner, especially the diffusion of the initially well-collimated sheet beam after undergoing multiple reflection. [Work supported by ONR.]

11:15

XX12. Ray acoustic analysis of plane-wave coupling into large open waveguides: Circular geometry. H. Shirai and L. B. Felsen (Polytechnic University, Route 110, Farmingdale, NY 11735)

Sound penetration into large enclosures can be modeled approximately by ray methods. When these enclosures are elongated so as to admit simulation by a waveguide, the interplay between rays and guided modes becomes important. To clarify the relevant phenomenology, we examine the prototype problem of a rigid semi-infinite baffled or unbaffled circular waveguide. Using the methodology of the geometrical theory of diffraction (GTD) and its modification to account for a guiding environment [L. B. Felsen and H. Y. Yee, J. Acoust. Soc. Am. **44**, 1028–1039 (1968)], the interior field for arbitrary plane wave incidence is constructed by edge diffracted-multiple reflected ray tracing, by modal expansion, and by physical optics (PO) applied to the aperture field. Stationary phase evaluation of the PO modal excitation coefficients establishes the connection with GTD, especially the mechanism of coupling from GTD into the modal fields. By reciprocity, the radiation problem can be handled in a similar way. Some conclusions are drawn for guides with noncircular cross section. [Work supported by ONR.]

11:30

XX13. Convergent transient acoustic fields from pulsed planar contracting ring sources. Stephen I. Warshaw (Physics Department, University of California Lawrence Livermore National Laboratory, P. O. Box 808, L-298, Livermore, CA 94550)

If a pulsed, radially symmetric planar source imbedded in a rigid baffle of infinite extent is described by a normal surface acceleration history of the form $a_n = \delta(t + r/v_s)$, where r is radial surface distance from the source center, v_s is the speed of uniform radial contraction, and the motion proceeds from $r = a$ to $r = 0$, then convergent and intersecting wave fronts can be produced in the nearfield of this source. Fields from this source have been calculated using techniques described previously [S. I. Warshaw, J. Acoust. Soc. Am. Suppl. 1 77, S60 (1985); and 79, S90 (1986); Proc. 12th ICA, Toronto, 1986] in the time domain. The calculation technique will be reviewed briefly and compared for contracting and expanding ring sources. Details of the time-evolving two-dimensional wave field will be presented in the form of contours of constant acoustic amplitude calculated at different times from source onset to propagation past the convergence zone. The various dynamically changing wave front patterns will be indicated and discussed. [Work supported by U.S. Department of Energy.]

11:45

XX14. Simple theoretical model for dynamic bulk modulus of elastometers and liquids. Jay Burns (Florida Institute of Technology, Melbourne, FL 32901) and Pieter S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

A free-volume theory for the temperature dependence of the dynamic bulk modulus is presented which is based upon the Cohen–Grest theory of the liquid–glass transition [M. H. Cohen and G. S. Grest, Phys. Rev. B **20**, 1077–1096 (1979)]. The theory leads naturally to the WLF method for constructing "master curves" for the bulk modulus analogous to those for shear and Young's moduli. It also provides for the transition at sufficiently high temperatures from the WLF regime to the range where the bulk modulus is dominated by an Arrhenius-type temperature dependence. The frequency dependence of the bulk modulus was developed from an approach used by Fuoss and Kirkwood in their theory of dielectric losses in polymers [R. M. Fuoss and J. G. Kirkwood, J. Am. Chem. Soc. **63**, 385–394 (1941)]. It leads to a hyperbolic tanget dependence of the storage modulus upon the logarithm of frequency and to a hyperbolic secant dependence of the loss modulus upon the log frequency. Experimental data on elastometers support the theory closely. [Work supported by ONR.]

FRIDAY MORNING, 12 DECEMBER 1986 — SALON F, 9:00 TO 11:15 A.M.

Session YY. Psychological and Physiological Acoustics VII: Binaural Hearing in Man and Animals

Erwin R. Hafter, Chairman

Department of Psychology, University of California, Berkeley, California 94720

Contributed Papers

9:00

YY1. Masking-level differences as a function of masker level: Revisited. William A. Yost (Parmly Hearing Institute/Loyola University, Chicago, IL 60626)

Measurement of the masking-level difference (MLD) has been suggested as a possible test for some types of hearing disorders. When MLDs are measured with hearing impaired patients, careful attention must be paid to the overall stimulus level and differences in sensitivity between the two ears. Studies done in the late 1960s proposed an explanation for the dependence of the magnitude of the MLD on masker level or interaural differences in masker level. This explanation assumes that additive internal noise present in the outer ear produces a significant contribution to the masking stimulus at the low signal frequencies typically used in studies of the MLD. The present paper will review this explanation and combine it with the predictions of the Durlach equalization–cancellation model of binaural analysis to fit the data from all the studies since 1948 that have investigated the MLD as a function of masker level or interaural masker level. In addition, data from a study using insert headphones will be described. The use of insert headphones, instead of the supra-aural headphones typically used to measure the MLD, should reduce the contribution to the masker of the additive internal noise present in the outer ear. [Work supported by the Air Force Office of Scientific Research, Life Sciences, and the National Institutes of Health, NINCDS.]

9:15

YY2. Envelope correlation preception. Virginia M. Richards (Department of Psychology, University of Florida, Gainesville, FL 32611)

The ability to determine whether simultaneously presented noise bands have correlated envelopes was examined. In each interval of a 2IFC paradigm, two bands of noise, centered at frequencies f and $f + \Delta f$, were presented. In one interval, the two noise bands had identical envelopes, while, in the other, interval envelopes were uncorrelated. Four frequency regions were tested: $f = 350$, $f = 1000$, $f = 2500$, and $f = 4000$ Hz. Percent correct discrimination was determined as a function of both f and noise bandwidth. The ratio $\Delta f / f$ ranged from 0.5–1. For $f = 350$ Hz, discrimination did not exceed chance. This agrees with the data of Schubert and Nixon [ONR Tech. Rep. NR **140**, 253 (1970)]. In contrast, it was found that for the larger values of f, observers could readilly discriminate between the correlated and uncorrelated conditions. In general, as the frequency separation (Δf) increased, performance decreased. The exception to this rule occurred at $f = 1000$ Hz. For two of the three observers, performance at moderate values of Δf exceeded performance at both small and large values of Δf. [Work supported by AFOSR and an NIH postdoctoral fellowship.]

9:30

YY3. Effect of spectral density and interaural parameters on profile analysis. Leslie R. Bernstein and David M. Green (Department of Psychology, University of Florida, Gainesville, FL 32611)

Detection of a change in spectral shape was investigated by incrementing the central, 1-kHz component of a multitonal complex. Complexes consisted of equal-amplitude components spaced logarithmically from 200 Hz to 5 kHz. Detectability was investigated as a function of the number of components in the complex and certain interaural parameters. Stimuli were presented diotically or in one of several dichotic configurations. As reported previously, for the diotic stimuli, performance improves as the number of components is increased from 3 to about 21; further increases yield *poorer* performance. The data from the dichotic conditions, however, are somewhat different from the diotic conditions. Like the data of Green and Kidd [J. Acoust. Soc. Am. 73, 1260–1265 (1983)], they suggest that although spectral analysis can be achieved using information across ears, performance is inferior to that obtained with diotic stimuli. However, *our* dichotic data do not reveal as large a decrement in performance relative to the diotic condition. Also, it appears that, with dichotic stimuli, performance may be affected by the presence of binaural cues. [Work supported by AFOSR.]

9:45

YY4. Discrimination of tonal complexes on the basis of which component is interaurally delayed. Raymond H. Dye, Jr. (Parmly Hearing Institute, Loyola University of Chicago, Chicago, IL 60626)

An experiment was undertaken to determine the extent to which information arising from interaural differences of time (IDTs) presented in one component of a complex is correlated with information arising from other components. Observers discriminated between multicomponent stimuli that consisted of the same spectral components but differed in terms of which component was interaurally delayed. Correlations were estimated by comparing discrimination d's to those predicted on the basis of the d's obtained for the detection of IDTs in each of the to-be-discriminated stimuli (versus diotic complexes). Discrimination d's were measured for three-component complexes consisting of 653, 753, and 853 Hz for the following conditions: 653_T-753-853 vs 653-753_T-853 Hz, 653_T-753-853 vs 653-753-853_T Hz, and 653-753_T-853 vs 653-753-853_T Hz, where T indicates an interaurally delayed component. In addition, d's were measured for discrimination of the three varieties delayed three-tone complexes from diotic complexes. Results show that interaural information arising from different components in these tonal complexes is highly correlated. [Work supported by the Air Force Office of Scientific Research, Life Sciences, the National Institutes of Health, NINCDS, and the National Science Foundation.]

10:00

YY5. Parametric study of onset effects in lateralization. Thomas N. Buell and Ervin R. Hafter (Department of Psychology, University of California, Berkeley, CA 94720)

When lateralization signals made up of trains of high-frequency clicks (n per train), optimum usage of the information carried in each click results in binaural thresholds that decrease at the rate $1/\sqrt{n}$. However, when the interclick interval (ICI) is short, the relation between threshold and n is best described by raising n to the exponent k [where $0<k<1$ and $k=f(\mathrm{ICI})$]. This implies that shorter ICIs produce greater dependence on information contained in the signal's onset, with $k=0$ meaning that no information is derived beyond the first click in the train while $k=1$ means that each click is equally effective. Based on discriminations of interaural differences of time (IDT), it will be shown that the relation between k and ICI is reasonably linear, ranging from a k of 0.0 for ICIs of 1–2 ms to k equal 1.0 for ICIs of about 11–13 ms. Interestingly, if the ICIs are made even longer, the k remains at 1.0 but the absolute performance grows worse. It is what one might expect if the internal jitter added to each sample of IDT was increased for widely spaced clicks but the observer was still able to integrate across the entire train. [Work supported by NINCDS.]

10:15

YY6. Localization of sound: Source discrimination and source identification. Brad Rakerd (Department of Audiology and Speech Sciences, Michigan State University, East Lansing, MI 48824) and W. M. Hartmann (Department of Physics, Michigan State University, East Lansing, MI 48824)

The ability of human listeners to localize sound sources can be measured by discrimination experiments, such as the minimum audible angle (MAA) paradigm, or by source identification experiments. In principle, these two kinds of experiments can be unified by a statistical decision theory model, incorporating a process whereby a sensory scale is referenced to an absolute scale. To test this idea we performed several kinds of free field discrimination and source identification experiments, both in the light and in the dark. Among our conclusions are the following: (1) The variance of the referencing process depends upon the width of the absolute scale required by the experiment. It doubles as the width increases from 2°–33°. (2) The MAA experiment, nominally a measurement of discrimination, probably includes a source identification component as well. In fact, our data in the light are consistent with a model in which the MAA experiment is treated exclusively as a source identification task. [Work supported by the NIH and NIMH.]

10:30

YY7. A weighted image model for binaural lateralization. Richard M. Stern and Andrew S. Zeiberg (Department of Electrical and Computer Engineering, Carnegie Mellon University, Pittsburgh, PA 15213)

A new model that predicts the subjective lateral position of bandpass binaural stimuli is described. As in other models, it is assumed that stimuli are bandpass filtered and rectified, and that the rectified outputs of filters with matching center frequencies undergo interaural cross-correlation. The results of these operations is a two-dimensional function $R(\tau,f)$ where f is center frequency, and τ refers to the delay parameter of the crosscorrelation operation. For many stimuli the modes of $R(\tau,f)$ form families of hyperbolae in the τ,f plane. Lateralization predictions are obtained from a weighted linear combination of the centroids along the τ axis of the trajectories of these modes. Individual trajectories receive greater weighting if they are *straighter* (i.e., more parallel to the f axis) and/or more *central* (i.e., closer to the f axis). The *straightness* factor enables the model to predict the preceived laterality of bandpass stimuli with combinations of interaural time delays and phase shifts, and amplitude-modulated stimuli with time delayed envelopes. This model is compared to other theories based on explicit processing of envelope information. [Work supported by NIH.]

10:45

YY8. The preception of complex echoes by an echolocating dolphin. Whitlow W. L. Au, Patrick W. B. Moore (Naval Ocean Systems Center, P. O. Box 997, Kailua, HI 96734), and Deborah A. Pawloski (SEACO, Inc., Kailua, HI 96734)

Dolphins echolocate with short broadband acoustic signals which have good time resolution properties. Received echoes are often complex, with many resolvable highlights or components caused by reflection of the incident signal from external and internal boundaries of a target and from different propagational modes within a target. A series of experiments was performed to investigate how dolphins perceive complex echoes. Echoes were produced by a microprocessor-controlled electronic target simulator which captured each emitted click and retransmitted the signal back to the animal after an appropriate delay time. The use of a "phantom" target allowed for precise control of the number of highlights, the time separation between highlights and the relative amplitudes of highlights in the simulated echoes. An echolocating dolphin was trained to perform a target detection task in the presence of masking noise using these phantom echoes. The properties of simulated echoes were systematically varied, and corresponding shifts in the dolphin's detection threshold were observed, allowing the inference of how the dolphin perceived the echoes. The dolphin performed like an energy detector with an integration time of approximately 290 μs.

11:00

YY9. Ear position affects the auditory space representation in the inferior colliculus of bats. Xinde Sun (Department of Biology, East China Normal University, Shanghai, People's Republic of China) and Philip H.-S. Jen (Division of Biological Sciences, The University of Missouri, Columbia, MO 65211)

Using free field acoustic stimulation conditions, the spatial response areas of single neurons and auditory space representation in the inferior colliculus (IC) of the big brown bat *Eptesicus fuscus*, was studied under different ear positions. A best frequency stimulus was delivered from the loudspeaker which was moved manually across the frontal auditory space of the bat to determine the response center of the neuron. At the response center, the neuron had its maximal directional sensitivity. The spatial response area of each neuron was determined for acoustic stimuli set at several intensity levels above the minimum threshold of the neuron. The spatial response areas of each IC neuron expanded unsymmetrically with the stimulus intensity. The size of the spatial response area was not correlated with the minimum threshold, best frequency, or recording depth of each neuron. Although each neuron had a point of maximal directional sensitivity, the point-to-point representation of the auditory space was not systematically organized. This representation was not correlated with the recording site of each neuron in the mediolateral, posteroanterior and dorsoventral axes of the IC. Both the response centers and spatial response areas of individual neurons varied with ear position but individual variations were complementary. It is likely that a mobile auditory space representation may provide the bat with versatility in maximizing the directional sensitivity of its echolocation system for accurate prey capture. [Work of Philip Jen supported by NIH.]

FRIDAY MORNING, 12 DECEMBER 1986 SALON E, 8:30 A.M. TO 12:18 P.M.

Session ZZ. Speech Communication VII: Speech Physiology; Speech Perception 1

John J. Ohala, Chairman

Phonology Laboratory, Department of Linguistics, University of California, Berkeley, California 94720

Contributed Papers

8:30

ZZ1. Nasality and nasal airflow in English. Sean Boisen (Department of Linguistics, University of California at Los Angeles, 405 Hilgard Avenue, Los Angeles, CA 90024)

This paper presents results from a general study of nasal airflow for a speaker of English. Nasal flow was recorded for approximately 1000 tokens in a carrier phrase, along with the audio output. The results support a hierarchical arrangement of vowels according to manifestation of nasal flow (in nasal environments) which parallels their oral impedance. This is consistent with acoustical studies of the effects of nasal coupling, as well as some perceptual studies, but differs in crucial ways from previous articulatory work. Other contextual segmental effects are observed, including a striking attenuation of the flow curve in final voiceless clusters. A revised definition of nasality, based on aerodynamic criteria, is advanced. Based on this definition, the circumstances under which nasality "spreads" was investigated, along with the flow characteristics of different segments. [Work supported by NIH.]

8:42

ZZ2. Nasal co-articulation in child and adult speech. James Emil Flege (Department of Biocommunication, University of Alabama at Birmingham, Birmingham, AL 35298)

The present study was motivated by the finding [Thompson and Hixon, Cleft Palate J. **16**, 412–420 (1979)] that the percentage of subjects showing emission of nasal airflow at the midpoint of prenasal vowels increased through childhood. An acoustical method was used to test the hypothesis that the domain of velar co-articulation increases with age. Adults and children aged 5 and 10 years (ten of each) produced /dVn/ and /nVd/ words eight times each at normal and fast speaking rates. Abrupt changes in nasalance scores (based on the ratio of the signal transduced by nasal and oral microphones separated by a thin plate) were used to determine what part of vowels (/i,ɪ,u/) were nasalized. There was no difference between groups in either the percentage of the vowel intervals in /dVn/ (78%) or /nVd/ (82%) that were nasalized, or in the absolute duration of the nasalized portion of the vowels (389 and 342 ms, respectively). This was true even after normalizing for between-group differences in vowel duration. These results suggested that the domain of velar co-articulation either does not change with age, or shows a mature pattern by the age of 5 years. In addition, the subjects in all three groups manifested the same substantial difference in nasalance scores in the context of /d–d/ (17.9) versus /n–n/ (58.9). Both adults and children showed significantly greater nasalance scores in the context of /i/ than /u/ or /ɪ/, presumably as the result of greater oral resistance to airflow in /i/. [Work supported by NIH.]

8:54

ZZ3. Laryngeal contributions to the difference between two types of whisper. Nancy B. Pearl (Speech and Hearing Sciences, University of Arizona, Tucson, AZ 85721) and Gerald N. McCall (Hearing and Speech Sciences, University of Maryland, College Park, MD 20742)

The laryngeal behaviors that contribute to perceptual differences (primarily loudness and noisiness) between whispers produced in a relaxed manner and in a forced manner are not well understood. To address this issue, we examined vocal fold configuration, glottal size, and supraglottal constriction during ten adults' productions of relaxed and forced whisper for three vowels in consonant–vowel syllable strings. Video recordings of the laryngeal structures were obtained through fiberoptic endolaryngoscopy. Results provided evidence that the differentiation of relaxed (quiet) and forced (loud) whisper was related to some extent to changes in supraglottal constriction (more for forced than relaxed whisper) but not to changes in glottal configuration or size. The failure of supraglottal constriction to occur for all instances of forced whisper implies that this strategy is an incomplete explanation for perceptual differences between whisper types. Simultaneous variations in both laryngeal and respiratory behaviors is a more tenable model, and deserves further consideration.

9:06

ZZ4. Interactions in the labial musculature during speech. John W. Folkins (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242), Raymond N. Linville (Department of Communicative Disorders, University of Pittsburgh, Pittsburgh, PA 15260), J. David Garrett (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242), and Carl Kice Brown (Department of Preventive Medicine and Environmental Health, University of Iowa, Iowa City, IA 52242)

Interactions in electromyographic activity of the upper and lower lips during speech were studied by manipulating the magnitude of bursts of

activity related to bilabial closure. Four pairs of electrodes were placed in the labial musculature in each of four normal-speaking young adults. Activity from both the upper and the lower lip was manipulated with the jaw free and then the procedures were repeated with the jaw fixed with a bite block. Manipulation of muscle activity usually resulted in positively correlated changes in activity recorded from the other three electrode pairs. Contrary to expectations, no significant negative correlations were found. The positive correlations may be due to (a) sensitivity of the electrode placements to the same motor units, (b) sensitivity of the electrode placements to motor units participating in common recruitment orders, or (c) functional interactions among muscles or muscle compartments that constrain them to maintain task-specific proportional relations. [Work supported by NINCDS.]

9:18

ZZ5. Vocal fold configuration associated with glottographic signals. Bruce R. Gerratt, David G. Hanson, and Gerald Berke (V.A. Medical Center, West Los Angeles, W126, Los Angeles, CA 90073 and UCLA School of Medicine, Los Angeles, CA 90024)

In recent years, glottographic techniques have been used more frequently to study vocal fold vibratory motion in clinical populations with voice abnormalities. However, some types of laryngeal pathology can result in deviant vocal fold movement, making interpretation of glottographic signals difficult. In the absence of other evidence, glottal events must be assigned to corresponding glottographic signals by inference usually from the study of normal phonation. This study was designed to confirm the relationship of simultaneously recorded photoglottographic (PGG) and electroglottgraphic (EGG) signals to vibratory movements of the vocal folds in the living human larynx. Photographs of the vocal folds were made using an 80-μs flash. The flash was recorded as an impulse on the PGG trace, indicating the location on the glottographic signals of the corresponding single-frame photograph displaying the glottal configuration. The instants at which the upper vocal fold margins first begin to open, separate completely, reach maximal lateral excursion, the lower margins first make contact, and close completely were documented. It appears that the timing of these events can be accurately identified in glottographic signals from normal and pathologic voices. [Work supported by NINCDS.]

9:30

ZZ6. Laryngeal compensation following sudden oral perturbation. Kevin Munhall, Anders Lofqvist, and J. A. S. Kelso (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

Recently, there have been a number of demonstrations of rapid goal-directed compensation in remote articulators when an articulator is perturbed unexpectedly during ongoing speech. Much of this research, however, has involved articulators that cooperate to produce a single vocal tract constriction (i.e., tongue–jaw, lips–jaw). In this experiment, spatially more separate articulators are examined that are simultaneously active in the production of a phonetic segment (i.e., the larynx and the lip–jaw complex). Three subjects produced the nonsense utterance "/ɩpɩp/ again" and the lower lip was unexpectedly loaded in the first vowel or during the closure for the first /p/ on a small proportion of the trials. Laryngeal behavior was monitored using transillumination and fiberoptic videotaping. In addition, intraoral air pressure just above the folds was monitored. Oral movements were recorded with a Selspot system. Acoustically, the first /p/ closure in "ɩpɩp" was shorter in duration and the voice onset times were longer in the perturbed trials. An examination of the laryngeal gestures during the production of the first /p/ revealed longer adduction durations in the perturbed conditions than in the control trials and a longer overall abduction/adduction cycle. The results will be discussed in terms of task-dependent articulatory coupling. [Work supported by NINCDS.]

9:42

ZZ7. Patterns of interarticulator phasing relations. Susan Nittrouer, Kevin Munhall (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511), J. A. S. Kelso (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Center for Complex Systems, Florida Atlantic University, Boca Raton, FL 33431), Betty Tuller (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Department of Psychology, Florida Atlantic University, Boca Raton, FL 33431), and Katherine S. Harris (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511)

The difficulty in identifying phonetic segments in the acoustic signal of speech is a commonly recognized problem. One proposed solution is that the integrity of individual segments is maintained by stable timing relations between the articulators. Recently, Kelso and Tuller [e.g., J. Acoust. Soc. Am. Suppl. 1 **77**, S53 (1985)] reported an invariant relation between jaw and lip gestures in various VCV contexts. Specifically, the onset of upper lip lowering for the intervocalic consonant was found to have a constant relation to the vowel-to-vowel jaw cycle in both time and position-velocity phase-plane domains. In this paper, we examine the additional effects of syllable affiliation and intervening consonant identity. The data show that the upper lip's lowering onset, relative to the position-velocity state of the jaw (phase angle), varied as a function of speaking rate and stress. In addition, the position of the syllable boundary (before or after the intervocalic consonant) and the consonant identity (*p* vs *m*) influenced the interarticulator phasing for some subjects. There was a general tendency for any condition that shortened the first vowel in the disyllables to produce an earlier onset of the upper lip relative to the jaw. However, the within-condition jaw cycle duration did not correlate with the within-condition variability in phase. The results are discussed in terms of the relationship between phonetic constraints and articulatory patterning. [Work supported by NIH.]

9:54

ZZ8. Effect of nerve section on spastic dysphonic voice: Implications for phonatory motor control. Thomas Shipp (Speech Research Laboratory, V.A. Medical Center, San Francisco, CA 94121) and Krzysztof Izdebski (V.A. Medical Center, Martinez, CA 94553)

The results of three laryngeal EMG studies on patients with spastic dysphonia (SD) show a similar pattern of inappropriate increased adductory muscle activity bilaterally during speech, resulting in phonation that is momentarily interrrupted or severely reduced in amplitude. Following these moments of "strained–strangled" phonation, the involved muscles resume activity at the previous levels and normal phonation is resumed. After temporary unilateral recurrent laryngeal nerve block, phonation is uninterrupted and no inappropriate buildup of activity is observed in muscles contralateral to the nerve block. A model of the SD process explaining these data identifies normal peripheral nerve and muscles, but abnormal afferent/efferent interaction at the brainstem level. Further, SD symptoms developed in post-surgical voice appear in patients with midline position of the paralyzed fold who produce voice at low $F0$ values.

10:06

ZZ9. An investigation of motor equivalence in the speech of children and adults. Bruce L. Smith and Ann McLean-Muse (Department of Communication Sciences and Disorders, Northwestern University, Evanston, IL 60201)

The concept of motor equivalence in speech production suggests that the contributions of various articulators can vary considerably across multiple productions of a segment, yet the end product is perceptually consistent. Since motor equivalence presumably represents a sophisticated ability in speakers, one purpose of the present study was to determine whether developmental trends could be oberved toward more frequent occurrence of motor equivalence with increasing age. A strain gauge transduction system was used to monitor inferior–superior upper lip, lower lip, and jaw movements produced by a group of adults and three groups of children, as they spoke normally and in two experimental conditions (bite block and fast rate). Assuming that negative correlations between pairs of articulators constitute evidence of motor equivalence, there was not a strong indication of motor equivalence in the speech of the children or the adults for a variety of comparisons. In addition, the children tended to show more negative correlations than the adults. All four groups also tended to exhibit negative correlations most frequently in the bite block condition. [Work supported by NINCDS.]

10:18

ZZ10. Increasing accuracy of coronal tongue imaging using ultrasound. Maureen Stone (Department of Rehabilitation, Building 10, Room 5D37, NIH, Bethesda, MD 20892), Thomas H. Shawker (Department of Radiology, NIH, Bethesda, MD 20892), Harold W. Tipton, Thomas L. Talbot, Alan H. Rich, and Joseph N. Lederman (Biomedical Engineering and Instrumentation Branch, NIH, Bethesda, MD 20892)

Coronal tongue images made using ultrasound have been limited by lack of precision in transducer placement. To improve precision and accuracy of data collection, especially in the coronal plane, a transducer holder and a head holder have been developed. The transducer holder maintains the transducer at a specified submental location in either the sagittal or coronal plane. Beam angulation is determined relative to the mandibular rami. An acoustic standoff prevents transducer interference with mandibular movement. The head holder consists of three cup-shaped cushions attached to the back of a dental chair which form a triangular support against the posterior head. A Velcro strap, placed around the head, is tightened by means of an eccentric. The head holder also contains a removable arm which can hold up to three strain gauge transducers allowing simultaneous lip and jaw measurements to be made. With increased instrumental accuracy, more subtle tongue shape comparisons can be made. The effects of coarticulation on midsagittal tongue groove, as seen coronally in /sVs/ and /pVp/ context will be presented at several transducer angles.

10:30–10:42

Break

10:42

ZZ11. Japanese quail categorize [d] across allophonic and talker variation. Keith R. Kluender and Randy L. Diehl (Department of Psychology, University of Texas, Austin, TX 78712)

Last fall, the ability of Japanese quail to categorize allophonic variants of [d] was reported. In those experiments, quail were trained to discriminate [dæs], [das], [dus], and [dis] from tokens with the same vowels following [b] and [g]. Quail then correctly categorized novel [d], [b], and [g] tokens with the vowels [ɛ], [ʌ], [ʊ], and [ɪ]. A possible criticism is that those novel nowel environments were not sufficiently novel, since [ɛ], [ʌ], [ʊ], and [ɪ] may be loosely characterized as lax versions of the vowels used in training. To address this point, we tested the same birds on four additional vowel environments: [ɝ], [e$^{\text{ɪ}}$], [o$^{\text{ɪ}}$], and [o$^{\text{ʊ}}$]. Performance with these new vowel environments essentially mimicked that with the former novel vowels. Additionally, whether the quails' ability to categorize [d] transfers to tokens from a second male talker was examined. One quail was tested using unreinforced novel [d], [b], and [g] tokens with the vowels [æ], [i], [ʌ], [ʊ], [e$^{\text{ɪ}}$], and [o$^{\text{ɪ}}$] from the second talker. The quail again pecked significantly more to novel unreinforced [d] tokens than to novel [b] and [g] tokens. These data support the view that there exists significant acoustic/aduitory commonalities among the class of [d] allophones across talkers that may serve as the basis of category formation for both humans and animals. [Work supported by NICHD.]

10:54

ZZ12. Perceptual confusions in the speech of tracheoesophageal talkers. Robert G. Haaf and Philip C. Doyle (School of Human Communication Disorders, Dalhousie University, Halifax, Nova Scotia B3H 1R2, Canada)

This project investigated the preception of speech stimuli produced by tracheoesophageal (TE) talkers. Four highly proficient TE talkers produced pre- and post-vocalic consonants within a 66-item word list comprised of CVC stimuli [M. S. Weiss and A. G. Basili, J. Speech Hear. Res. **28**, 294–300 (1985)]. Stimuli were representative of all manner classes (stops, fricatives, affricates, nasals, and semi-vowels). Twelve normal-hearing young adults served as listeners. Listeners were trained in IPA transcription and had no, or minimal exposure to alaryngeal speech. Listeners were presented TE stimuli via headphones and were required to record their responses using broad phonetic transcription. An open-response paradigm was employed. All talker stimuli were randomized. Confusion matrices for each talker were prepared and overall intelligibility for each talker was determined. Analyses of the data were conducted for individual talkers, and by talker group. Categories of perceptual confusions were analyzed across manner class by phonetic context (pre- and post-vocalic). Voicing distinctions were also quantified by manner and phonetic context. Predominant perceptual confusions will be discussed as related to the unique alaryngeal speech production method used by this clincal population.

11:06

ZZ13. Speech sound identification influenced by adjacent "restored" phonemes. John J. Ohala and Deborah Feder (Phonology Laboratory, Department of Linguistics, University of California, Berkeley, CA 94720)

When listeners' identifications of speech sounds are influenced by adjacent sounds, do they operate only on the physical phonetic characteristics of these sounds or could they also use just their linguistic identity? We tested this by leading subjects to restore or induce the noise-obliterated medial consonant in VCV utterances by first presenting them with several prior utterances where this medial consonant was consistently the same, either a /b/ or /d/. Included as V1 were synthetic vowels from the /i–u/ continuum. As expected, more /u/s were identified out of this continuum in the environment of physically present /d/s than /b/s [Ohala *et al.*, J. Acoust. Soc. Am. Suppl. 1 **64**, S18 (1978)]. But restored /d/s and /b/s had the *same* effect (though with slightly less intensity) thus indicating that the influence of context need not operate only via physical phonetic features. We discuss the implications of this finding for "direct realist" theories of speech perception and for claims of the existence of invariant phonetic elements. [Supported by the Cognitive Science Program, UCB.]

11:18

ZZ14. Perceptual restoration of music. Lucinda A. DeWitt and Arthur G. Samuel (Department of Psychology, Box 11A Yale Station, Yale University, New Haven, CT 06520)

A number of studies have investigated the perceptual restoration of deleted speech sounds. The present study explored a musical analog to the speech case. Several experiments investigated whether or not listeners restore notes that have been deleted and replaced by matched white noise. The experiments used Samuel's [JEP: General **110**, 474–494 (1981)] "added–replaced" methodology to separate perceptual restoration from any higher level decision factors. In one experiment, we compared restoration of phonemes in words and notes in faimilar melodies; a single-note control condition was also run. Using the measure of percpetual restoration, words produced stronger restoration of their components than melodies did; the melodies produced stronger restoration than the controls. In other experiments, we have investigated the role that familiarity plays in perception of melodies, and the influence of melodic and rhythmic structure. We will consider the implications of the results for theories of speech and music perception. [Work supported by AFOSR and NSF.]

11:30

ZZ15. Effects of acoustical context on perceived vowel quality. Helen Pattison, Roy B. Gardner, and C. J. Darwin (Laboratory of Experimental Psychology, University of Sussex, Brighton BN1 9QG, England)

Vowels along an /ɪ/–/e/ continuum with added energy at a harmonic frequency near to F are embedded in sequences of tones and their phonetic quality assessed by the position of the phoneme boundary. If the sequence of tones has the same frequency as the added energy, then its contribution to vowel quality decreases as the number of tones preceding the vowel is increased, and is greater if the tone sequence continues after the vowel, indicating a greater tendency for the added tone to be perceptually segregated from the vowel. A similar effect is observed if the vowel is embedded in a sequence of tones which *descends* in frequency (by whole tones, or by half-harmonic steps), coinciding with the vowel at the harmonic frequency. However, if the tone sequence *ascends* in frequency, then the added tone *does* alter the perceived vowel quality, indicating that the tone sequence has not captured the embedded tone. In a further experiment a single tone at various frequencies preceded the vowel by various time intervals. The experiment showed that the difference between ascending and descending series was due to the cumulative individual effects of the two tones that immediately preceded the vowel, rather than to the configuration of the series. [Work supported by SERC.]

11:42

ZZ16. Perceptual compensation for effects of transmission channel characteristics on vowel quality. Anthony J. Watkins (Department of Psychology, University of Reading, Reading RG6 2AH, England), Helen Pattison, and C. J. Darwin (Laboratory of Experimental Psychology, University of Sussex, Brighton BN1 9QG, England)

The phonetic quality of front vowels along a synthetic /ɪ/–/e/ F1 continuum can be influenced by passing the vowels through a filter that has a sloping characteristic in the F1 region. The shift in phoneme boundary that occurs for different filter slopes was measured and whether the influence of various conditions listeners can compensate for the characteristics of the transmission channel was examined. These conditions include placing the vowel in a consonantal context and preceding the vowel with a carrier phrase that has been subjected to the same filtering as the vowel. Experiments have also been performed on the preception of natural vowels in isolation and in a consonantal context. [Work supported by SERC.]

11:54

ZZ17. Effects of lexical stress placement on auditory word recognition. Louisa M. Slowiaczek (Department of Psychology, Loyola University of Chicago, 6525 North Sheridan Road, Chicago, IL 60626)

Research was conducted examining the role of lexical stress in auditory word recognition. In a series of experiments, isolated spoken words were presented to subjects with correctly produced stress pattern (CS) or incorrectly produced stress pattern (IS). In three separate experiments subjects were asked to identify the CS and IS items in varying amounts of white noise, make a lexical decision or monitor for any items that were nouns. The results revealed no effect of stress replacement on identification of words in noise, but revealed a significant effect of stress placement on lexical decision and noun monitoring response times. The role of lexical stress information in auditory word recognition will be discussed in light of these results. [Work supported by NIH.]

12:06

ZZ18. Listener biases in diagnostic intelligibility testing. Caldwell P. Smith (Electromagnetic Sciences Division, Rome Air Development Center, RADC/EEV, Hancom AFB, MA 01731)

In the course of conducting numerous diagnostic rhyme tests for intelligibility evaluations of many voice processors over the past several years, several retests were run as checks of reliability. Retests involved assigning new identification numbers to test recordings and resubmitting them for evaluation by presentation to eight-member listener crews. Retests were interspersed with other (new) tests and were usually run about a week after the first presentation. Overall intelligibility scores from the two test administrations have been found to be in close agreement. However, analysis of differences in paired diagnostic scores from the two tests has revealed significant effects of listener biases. The pool of listeners consisted of subjects screened for normal hearing and having extensive familiarity and experience with the test materials and procedures. When the second test was conducted with the identical listener crew, none of the 54 feature scores or the 18 speaker scores differed significantly. When all eight listeners were different, 18 of the 54 feature scores and three of the 18 speaker scores differed significantly ($a = 0.05$). For six cases in which the number of common listeners ranged from one to six, the number of diagnostic scores that differed significantly in the two tests tended to vary inversely with the number of listeners in common ($r = -0.77$). This finding suggests that it cannot be assumed that two eight-member listener crews of equivalent competence will have equivalent performance in discriminating among phonetic events having minimal contrast.

FRIDAY MORNING, 12 DECEMBER 1986 — SALON G, 8:30 A.M. TO 12:00 NOON

Session AAA. Underwater Acoustics IX: Signal Processing and Underwater Acoustics

Peter F. Worcester, Chairman
Scripps Institute of Oceanography, University of California, La Jolla, California 92093

Contributed Papers

FRIDAY AM

8:30

AAA1. Constant beamwidth receiving arrays for broadband detection and power estimation of signals. Joseph Lardies and Jean-Pierre Flenner (Laboratoire d'Acoustique, de Métrologie et d'Instrumentation, Université Paul Sabatier, 38 Rue des 36 Ponts, 31062 Toulouse Cedex, France)

The detection at low frequencies and power estimation of broadband signals may often be accomplished with an array of sensors. Four beamformers are studied; the conventional end-fire beamformer, and the maximum-likelihood estimator, the maximum-entropy estimator, and a new broadband constant beamwidth beamformer. It is shown that it is impossible to detect and estimate the power of small signals at low frequencies by the conventional end-fire array, by the maximum-likelihood, or by the maximum-entropy methods. An algorithm with superior low-frequency performance is studied. The proposed method estimates the signal powers from the correlation between sensors. We find that the beamwidth of the processor is remarkably insensitive to frequency variations and is under operator's control. This technique may detect and estimate the power of signals, at low frequencies, which could not be found using the other techniques.

8:45

AAA2. The influence of correlogram spread and wander on a broadband cross-correlation system. P. Bilazarian (Raytheon Company, Submarine Signal Division, 1847 West Main Road, Portsmouth, RI 02871)

Scattering or time-varying mechanisms in the oceanic medium can affect the performance of a passive broadband cross-correlation system. These mechanisms include internal waves, sea-surface motion, and angular spreading caused by rough boundaries. The system involves three horizontal, collinear sensors and source location parameters are determined from time-delay estimates obtained by cross-correlation techniques. Tests for measuring the amount of spread or wander in correlograms with sufficiently high signal-to-noise ratios are presented. These tests are useful for determining the magnitued of the combination of scattering and time-varying effects in correlogram data. Spread is measured by comparing the average main-lobe width of stabilized correlograms over a period of time to the width of an ideal, "frozen-ocean" correlogram. Wander is estimated by a comparison of the standard deviation of correlogram peak location values over a period of time to the average main-lobe width of these correlograms. Examples of the application of the tests to simulated data are discussed. Finally, there is an identification of several correlogram types that can be associated with large amounts of spread and wander.

9:00

AAA3. Simulation of side-scan sonar. Aubrey L. Anderson and Shufa Dwan (Department of Oceanography and Geodynamics Research Institute, Texas A&M University, College Station, TX 77843)

Acoustic remote sensing is becoming widespread in oceanographic measurement programs. Acoustic systems for such measurements include seafloor subbottom profilers, inverted echo sounders, tomography systems, and side-scan sonars. Numerical simulation of such systems can provide valuable information for system design and for analysis of data from an operating system. A long-range side-scan system currently under development will operate in deep water over a frequency range from 12–150 kHz and will provide side-scan images, bathymetry, and seafloor reflectivity data. Simulations of the system have been performed. These simulations provide insight into the influence of various system parameters and dispalys on usefulness of the system as a tool for the seafloor geologist. This paper describes the models used in the simulations and illustrates their use in performance prediction for this scientific data acquisition system.

9:15

AAA4. Optimum time domain signal transmission and source location in a waveguide: Ideal wedge waveguide. Saimu Li[a] and C. S. Clay (Geophysical and Polar Research Center, University of Wisconsin, Madison, WI 53706)

Transmissions in an ideal wedge waveguide in air are described. The walls are rigid. The Biot–Tolstoy exact wedge solution is used. The diffraction arrival from the apex is not present. Signals transmitted in a waveguide are distorted by the multiple arrivals. A waveguide matched filter has an impulse response that is the time reverse of the impulse response of transmission between a source and receiver. This filter is an optimum filter for the reduction of transmission distortion. Since the waveguide matched filter is sensitive to source and hydrophone locations, the waveguide can be used to locate a source that transmits an unknown signal. [Supported in part by the Shangdong College of Oceanography People's Republic of China, the Geophysical and Polar Research Center of the University of Wisconsin, and NORDA.] [a] On leave from the Shangdong College of Oceanography, Qingdao, People's Republic of China.

9:30

AAA5. Optimum time domain signal transmission and source location in a waveguide: Ideal waveguide experiments. C. S. Clay and Saimu Li[a] (Geophysical and Polar Research Center, University of Wisconsin, Madison, WI 53706)

Experimental measurements that were made in an ideal waveguide in air are reported. The experiments confirmed the theoretical technique. Signals transmitted in a waveguide are distorted by the multiple arrivals. A waveguide matched filter has an impulse response that is the time reverse of the impulse response of transmission between a source and receiver. This filter is an optimum filter for the reduction of transmission distortion. The waveguide matched filter is sensitive to source and hydrophone locations, and it can be used to locate a source that transmits an unknown signal. Each signal channel is match filtered for a trial source location. Pairwise cross correlations have a maximum when the trial location is the same as actual source position. [Supported in part by the Shangdong College of Oceanography, People's Republic of China, the Geophysical and Polar Research Center of the University of Wisconsin, and NORDA.] [a] On leave from the Shangdong College of Oceanography, Qingdao, People's Republic of China.

9:45

AAA6. Use of rectangular vandermonde matrix decomposition to resolve damped sinusoidal oscillation components. T. L. Henderson, R. S. Bailey, and S. G. Lacker (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029)

Signals that are sums of decaying sinusoids arise frequently in passive and active sonar applications, both as spontaneous acoustic transients and as echoes from elastic objects. When the oscillations are highly damped, identification of component frequencies and damping factors is more amenable to least-squares Prony analysis than to Fourier analysis, and is particularly suited to a related technique that we call "rectangular Vandermonde matrix decomposition." Our results indicate that the Vandermonde method is robust in the presence of noise, but not in the presence of distortions created by ideal bandpass filters. On the other hand, analysis of simulated replica-correlated echoes of FM pulses bounced off an object having a pair of heavily damped resonances give different results. Surprisingly, the Vandermonde method identified the resonant frequencies even without replica correlation, despite the fact that the analyzed waveform looked not at all like a sum of decaying sinusoids. [Work supported by the ONR.]

10:00

AAA7. A time domain formulation of the absorption limited transient parametric array. Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

The impulse response of the absorption limited transient parametric array is developed using a time-dependent Green's function approach. A simple closed-form expression is developed for the farfield spatial and time-dependent impulse response of the parametric array. Farfield pressures are expressed as a time convolution of the impulse response and a source function. Numerical results for the farfield pressures resulting from a Gaussian-modulated carrier are presented to illustrate the spatial and time-dependent characteristics of the field. Additional numerical results are presented for the farfield pressures resulting from the pulsed excitation of a simple model of an ultrasonic transducer. These latter results are compared to existing experimental data and noted to be in general agreement. [Work supported by ONR.]

10:15

AAA8. Localization of geophone stations by maximum-likelihood estimation. Henrik Schmidt, Michael Snoek, and Paul van Alphen (SACLANT ASW Research Center, I-19026 La Spezia, Italy)

Reliable interpretation of seismic data in ocean bottom surveys or crust and upper mantle investigations requires precise knowledge of the position of the seismic sensors. Special problems arise when operating with free-falling ocean bottom seismometers (OBS). Since the ship's position and its towed signal sources (air gun, uniboom) or explosives are fairly well known based on high-precision navigation systems, one can determine the geophone positions indirectly from their response to the source excitation. Here, a newly developed source parameter estimation code [A. B. Baggeroer, W. A. Kuperman, and H. Schmidt, J. Acoust. Soc. Am. Suppl. 1 **79**, S56 (1986)] is first used to determine the performance bounds for a localization based on seismic interface waves. Then, an actual shear speed profile is determined by inversion of a new data set obtained using explosive sources, and a maximum-likelihood estimation of the geophone position is performed. By comparing the results with the actual experimental geometry, it is demonstrated that in spite of the fairly long wavelength of the interface waves, a reasonable resolution can be achieved by such a technique.

10:30

AAA9. Simulations of matched-field processing in a deep-water Pacific environment. Michael B. Porter, Ronald L. Dicus, and Richard Fizell (Code 5120, Naval Research Laboratory, Washington, DC 20375)

Matched-field processing is a signal processing technique for arrays in which field vectors for assumed source positions (range and depth) are substituted for plane-wave steering vectors in conventional linear and nonlinear beamformers. The field vectors are computed by standard acoustic field models (FFP, normal mode, etc.) which take into account propagation effects in an oceanic waveguide. The output is an ambiguity surface over possible source positions in which a peak is expected at the true source position. Accuracy of the computed fields is limited in large part by our knowledge of the environment. This environmental mismatch causes degradation in localization performance, sometimes leading to large errors in estimation of source position. In order to assess the significance of this effect, simulations were performed in which a measured field is synthesized using a slightly different environmental model from that used for the steering vectors. The differences were introduced to simulate expected errors in sound-speed profile, sediment thickness, and elastic wave speed. Calculations were made for a cw source operating at 10 Hz and depths of 25 and 250 m in a 500 m-deep ocean. The receiver was a 16-element vertical array at ranges of 25 (shadow zone) and 100 km (second convergence zone). A typical Pacific sound velocity profile was assumed. The bottom was modeled by a thin (50-m) sediment layer overlying an elastic subbottom. Degradation in localization performance due to environmental mismatch will be discussed both quantitatively and qualitatively.

10:45

AAA10. Preliminary results of matched-field localization using a vertical array in the Tufts Abyssal Plain. R. G. Fizell (Code 5120, Naval Research Laboratory, Washington, DC 20375)

PACIFIC ECHO was an experiment conducted jointly by the Naval Research Laboratory (NRL) and Defence Research Establishment, Pacific (DREP) in May–June 1986. Vertical array measurements of a 15-Hz cw signal projected by the NRL Mk-VI source, towed at a depth of 100 m on a circular arc of radius 3 nmi, were analyzed with matched-field processing. The DREP array was 675 m long with the top hydrophone at 400-m depth. Matched-field processing between theoretical and measured fields was accomplished by linear correlation and by a nonlinear maximum-likelihood method (MLM) estimator. Theoretical fields were computed using a normal-modes program, the measured sound velocity profile, and an assumed thick-sediment bottom. Successful localization of the source was achieved with both estimators. Sidelobes produced by the MLM estimator were significantly below the main peak but, for the linear estimator, sidelobes were sufficiently large to be taken as false targets.

11:00

AAA11. Sensitivity of the matched-field source localization technique in shallow water to mismatch of geoacoustic parameters. Donald R. Del Balzo (NORDA Code 244, NSTL, MS 39529), Christopher Feuillade, and Mary M. Rowe (ODSI Defense Systems, Inc., 6110 Executive Boulevard, Rockville, MD 20852)

The accuracy of matched-field techniques for source localization in a shallow-water waveguide is dependent upon realistic modeling of the geoacoustic environment. A study was conducted to investigate the sensitivity of source localization to erroneous estimates of the geoacoustic properties (sound speed, density, and attenuation) of the sediment. A range-independent normal-mode computer program was used to calculate acoustic fields from a midwater source received on a vertical array of 21 hydrophones spanning the water column. Errors in estimates of range and depth are presented as a function of mismatch of each geoacoustic property.

11:15

AAA12. Wave-height fluctuation effects on matched-field detection and localization in shallow water. Donald R. Del Balzo (NORDA Code 244, NSTL, MS 39529), Christopher Feuillade, and Mary M. Rowe (ODSI Defence Systems, Inc., 6110 Executive Boulevard, Rockville, MD 20852)

Sea surface wave-height fluctuations cause a time-dependent mismatch in environmental conditions and therefore effect the detection and localization performance of a matched-field processor. A sensitivity study was conducted to examine this mismatch phenomenon for an idealized, range-independent, Pekeris channel of 100-m depth, with a 150-Hz source, using a centrally positioned vertical array of 21 hydrophones spanning 50% of the water column. Variations in the sea surface height of $+3.5$ m were considered as an extreme, but realistic, case, and the output signal-to-noise ratio (SNR) and predicted range and depth of the source were determined from a series of range-depth maximum-likelihood ambiguity surfaces. Surface height variation caused a systematic error in range estimation, such that when the depth was increased due to a wave crest, the target range appeared shorter than the actual range. The opposite held for a wave trough. The corresponding calculations of target depth were consistently biased towards greater depth. Calculations indicated that small (1%) variations in surface height can cause a loss of up to 15 dB in detection performance at a single time. However, when the computations are properly normalized and then averaged over time throughout a complete cycle of the wave-height variations, the resulting detection is dominated by the zero wave-height maximum-likelihood surface, and localization estimates based upon the position of the main peak are unambiguous.

11:30

AAA13. Normal-mode filtering with orthogonal functions to avoid mode leakage. Grayson H. Rayborn (Department of Physics and Astronomy, University of Southern Mississippi, Hattiesburg, MS 39406-5165 and NORDA, NSTL, MS 39529), George E. Ioup, Juliette W. Ioup (Department of Physics and Geophysical Research Laboratory, University of New Orleans, New Orleans, LA 70148 and NORDA, NSTL, MS 39529), and Janet C. Carr (NORDA, NSTL, MS 39529)

Propagation of sound in shallow water is commonly analyzed in terms of trapped normal modes. The process of normal-mode detection by adjustment of the response of individual hydrophones in a vertical array to match the pattern of a particular normal mode is normal-mode filtering. Despite the nonorthogonality of the normal modes in the water column, normal-mode filtering is based on the assumption that the modes are orthogonal. The error induced by this assumption is known as leakage. It has been estimated to range from 3% to 10% of a measured mode. In this paper, the error involved in treating the modes as orthogonal for a variety of bottom types, water depths, and source frequencies is analyzed, and a method for avoiding the error is demonstrated. Density and sound-speed ratios are selected from models fitted to experimentally measured values. To render the results indicative of a broad range of water depths and frequencies, use is made of dimensionless variables. Leakages as large as 19% are calculated.

11:45

AAA14. Application of array processing to active noise cancellation in underwater acoustics. W. S. Gan (Acoustical Services Pte Ltd., 29 Telok Ayer Street, Singapore 0104, Republic of Singapore)

In this paper, array processing is applied and extended to wide, distributed noise sources. Array processing increases the speed of computation of FFT and Green's function. For narrow-band noise, a linear array of sensors is used. The array propagation vector and the weight vector are derived. In order to obtain optimal solution for noise cancellation, four correlation matrices are derived. Adaptive filters and Widrow's mean-square error algorithm are used. The optimal "Wiener" solution for the Wiener–Hopt equation in matrix form is obtained. For broadband noise, Wiener linear minimum mean-square error filtering is also used. The spectral density function of the noise component to be cancelled is derived. The broadband Wiener solution which is a generalization of the narrow-band case is obtained.

FRIDAY MORNING, 12 DECEMBER 1986 — SALON H, 8:30 A.M. TO 12:00 NOON

Session BBB. Underwater Acoustics X: Scattering from the Seafloor and Ice Canopy

Timothy K. Stanton, Chairman
Department of Geophysics, University of Wisconsin, Madison, Wisconsin 53706

Contributed Papers

8:30

BBB1. An all-frequency solution of sound scatter by hard angular surfaces. I. Tolstoy (Knockvennie, Castle Douglas, Southwest Scotland, United Kingdom)

In estimating plane sound-wave scatter by hard bodies of simple prismatic cross section or by corrugated surfaces with angular (i.e., sawtooth or facetted) profiles it is possible to replace the actual bodies by suitable systems of single (infinite) wedges and corners plus line sources (at the vertices) plus images thereof. This procedure differs fundamentally from approximate superpositions of rigorous wedge/corner solutions used by other writers in that *all* orders of multiple scatter interaction are taken into account *via* the standard self-consistent algorithm. An infinite series representation of the field was obtained; the coefficients of which are determined by an infinite system of simultaneous linear equations, which appear to be soluble in a number of cases. The resulting solution for the total field is convergent everywhere and satisfies all the boundary conditions. In the case of an angular depression in a plane, or in the troughs of a corrugated surface with a sawtooth profile, a simple, highly accurate solution is obtained for all frequencies. [Work supported by ONR.]

8:45

BBB2. A study of bathymetric scattering using simulation techniques. R. N. Baer, J. S. Perkins, and D. H. Berman (Code 5160, U.S. Naval Research Laboratory, Washington, DC 20375-5000)

Seafloor scattering can significantly affect the acoustic field. The statistics of ocean floor scattering were simulated by using the split-step parabolic equation with an ensemble of possible bathymetries in a manner analogous to that of Tappert and Nghiem-Phu [J. Acoust. Soc. Am. Suppl. 1 77, S102 (1985)]. The bathymetry was taken to have a Gaussian spectrum. Situations with various horizontal correlation lengths and rms roughness were examined. Different acoustic sources were also considered. An ensemble of 100 to 200 parabolic equation runs was found that in general was sufficient to give predictions of the mean and standard deviations of the acoustic field. The ensemble-mean distribution differs significantly from both an individual ensemble member and field with the average (flat) bathymetry. In the convergence zone, the variance of the acoustic field correlates with the mean transmission loss and it decreases with decreasing rms roughness or increasing horizontal correlation length. Outside the convergence zone, the variance is high even for relatively smooth cases. [Work supported by NRL and ONR.]

9:00

BBB3. Backscatter at a model of the Arctic ice canopy. Kevin R. Johnson, Patrick L. Denny (Physics Department, Naval Postgraduate School, Monterey, CA 93943), Ken J. Reitzel, and Herman Medwin (Ocean Acoustics Associates, Pebble Beach, CA 93953)

The mechanisms which may cause low-frequency sound to backscatter from the Arctic ice canopy include rough surface backscatter, diffraction at leads, and reradiation of flexural waves. These phenomena have been studied by observing underwater sound pulses backscattered from an acrylic model of the Arctic ice floating on water. The flexural wave velocity and the acoustic roughness are accurately scaled and the ρc impedance contrast is approximately modeled by the selection of the acrylic material and the dimensions of the plate and roughness elements. The 3-mm-thick plate backscattering at 50 kHz then represents an Arctic ice cover 1.5 m thick and about a kilometer in extent, backscattering 100-Hz sound. The physical contributors of backscatter from the Arctic surface are studied for models of plane ice, an Arctic ice ridge, edges of leads, and a rubble field of ice. [Work supported by ONR.]

9:15

BBB4. Forward scatter at a model of the Arctic ice canopy. Patrick L. Denny, Kevin R. Johnson (Physics Department, Naval Postgraduate School, Monterey, CA 93943), Ken J. Reitzel, and Herman Medwin (Ocean Acoustics Associates, Pebble Beach, CA 93953)

When low-frequency underwater sound "reflects" from the Arctic ice cover, not only will it be reflected from the plane, and specularly scattered from roughness elements, but it will also be diffracted at leads and reradiated from flexural waves in the ice. In the case of near-grazing forward scatter from closely spaced roughness elements there may also be a coherent forward scattered boundary wave detectable near the under-ice surface. These phenomena have been studied in an anechoic tank by pulse transmission from an underwater point source to a large floating acrylic model of the Arctic ice. Scaling of the flexural wave velocity $B(\omega h)^{1/2}$ at frequency ω and ice thickness h is accurately achieved because B, for the acrylic model, is well within the range of typical values in arctic ice. Scaling of the sound scattered from roughness elements is also accomplished in terms of frequency and the length dimension. Consequently, the laboratory model at frequency 50 kHz represents 100-Hz sound encountering a typical Arctic canopy which is $1\frac{1}{2}$ m thick and about a kilometer in extent.

The physical contributors to the gross "reflection coefficient" are studied for models of a plane ice layer, an Arctic ice ridge, edges of leads, and a rubble field of ice. [Work supported by ONR.]

9:30

BBB5. Laboratory-scale simulation of under-ice reflectivity at low frequencies. T. McLanahan (Naval Air Systems Command, Washington, DC), O. I. Diachok, and S. C. Wales (Code 5120, Naval Research Laboratory, Washington, DC 20375)

Under-ice reflectivity is dominated by sea ice ridges, long, randomly spaced, randomly oriented rubble piles of broken ice. The effective low-frequency shear and compressional velocities of these formuations have been estimated to be between 1100–1700 and 2700–3400 m/s, respectively (Diachok, Proceedings International Congress on Acoustics, 1986). To be consistent with these estimates, Lucite, a readily available and easily machined material, for which $V_s = 1260$ m/s and $V_p = 2600$ m/s was used to construct laboratory scale analogs of sea ice ridges situated on a thin plate. Both parallel and randomly oriented ridge models were constructed, and the effects of ridge elasticity and orientation on under-ice reflectivity (amplitude and phase) were examined for $0.67 < ka < 6$ (k = wavenumber and a = mean depth). Initial reflection coefficient data at the lowest frequencies are approximately consistent with Twersky's predictions for reflectivity from both hard and soft half-cylinders on a soft boundary. Higher frequency results, including phase versus grazing angle, will also be reported, and implications for under-ice propagation modeling will be discussed.

9:45

BBB6. Arctic low-frequency propagation loss from shallow cracks in ice plates. Jacques R. Chamuel (Sonoquest/Advanced Ultrasonics Research, P.O. Box 153, Wellesley Hills, MA 02181) and Gary H. Brooke (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

At the 111th Meeting of the Acoustical Society of America, the authors reported on the effects of shallow cracks in floating ice plates on lowering the flexural wave dispersion curve creating discrepancies in plate thickness determination. In this paper, the effects of shallow cracks in sea-ice plates on low-frequency propagation loss in the water waveguide underneath the ice are demonstrated using laboratory ultrasonic models. Air-filled shallow cracks extending near vertically in the ice plate present sharp impedance discontinuities to the low-frequency water acoustic waves coupled to the ice plates. Upward refraction in arctic water increases the interaction of the acoustic waves with the cracks causing distributed backscatter along the water waveguide. The importance of the cracks increases as the effective water waveguide depth is decreased. Experimental laboratory results are presented for shallow water waveguide conditions. The new findings may explain the very high backscatter levels and high propagation losses of low frequencies observed in the Arctic. [Work supported by DREP.]

10:00

BBB7. Geoacoustic scattering from seafloor features in the ROSE area. Martin E. Dougherty (Department of Geology and Geophysics, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A strong "refraction branch diffraction" has been observed on ocean bottom hydrophone data from the Rivera Ocean Seismic Experiment (ROSE). Modeling of the interaction between seismic/acoustic energy and the ocean bottom has shown that this arrival could be caused by scattering from common seafloor features. Seismic/acoustic wave propagation was computed for models of marine seismic refraction lines using a 2D heterogeneous formulation of the elastic wave equation. These models demonstrate that a significant amount of energy is diffracted from seafloor structures such as hills and valleys when both the direct water wave and the refracted compressional wave (traveling through the upper crust) are incident upon a structure. Since much of the diffracted energy travels through the crust, the diffracted arrivals appear on the refraction branch of the seismograms at large ranges. Energy partitioning at the seafloor of the two incident wave types produce both compressional (P) and shear (S) diffracted phases in the upper oceanic crust. The large models used also clearly demonstrate the existence of phases which are theoretically possible but rarely identified in marine seismic data such as the pseudo-Rayleigh wave and the P and S interference head waves.

10:15

BBB8. An nth order solution of the Neumann series in scattering from rough interfaces. M. F. Werby and Richard Keiffer (Naval Ocean Research and Development Activity, NSTL, MS 39529)

Scattering from rough interfaces has been treated by the integral representation of the Helmholtz equation. This equation involves surface integrals and is well suited for rough interfaces which are piecewise continuous. The well-known Kirchoff approximation involves replacing the unknown surface pressure field and its gradient by the incident field as a first-order approximation to the integral equation. This leads to an expression which involves direct integration and will be noted to represent the first term in the Neumann solution of an integral equation of the second kind. Physically, it corresponds to one encounter of the field with the interface. It can be shown that the nth solution corresponds to n encounters with the interface. We derive expressions for the nth solutions of the Neumann series. The expressions are exact for each order for the three-dimensional case and include an out-of-plane scatter.

10:30

BBB9. Single backscatter reverberation. Johanan L. Codona and Richard S. Patton (AT&T Bell Laboratories, Whippany Road, Whippany, NJ 07981)

By making use of Green's function techniques, a self-consistent, nonperturbative, single backscatter model for acoustic reverberation in the ocean was constructed. The model accounts for multiple forward scattering on both the outgoing and return paths. The model can be made to use any one-way wave equation: e.g., the standard parabolic equation or the Thomson–Chapman high-angle equation. When the standard parabolic equation is used, the technique lends itself to a numerical implementation that is much faster than competing schemes. While the model is particularly useful at low frequencies, its validity does not depend on frequency. The model's numerical speed and versatility suggest its use for the study of low-frequency reverberation from rough ocean surfaces, deterministic and random variations in the ocean volume, and range dependence in the bottom (both bathymetric and compositional).

10:45

BBB10. Sound propagation through random currents using parabolic approximations. R. I. Brent, M. J. Jacobson, and W. L. Siegmann (Department of Mathematical Sciences, Rensselaer Polytechnic Institute, Troy, NY 12180-3790)

The influence of random, horizontal, depth-dependent ocean currents in the parabolic approximation is studied. Emphasis is placed on received acoustic intensity. Parabolic equations including currents, derived recently by our group, are examined using perturbation and asymptotic methods. Expressions are derived for the mean and standard deviation of intensity. The development assumes that the random current is wide-sense stationary. For convenience, an exponential-cosine autocorrelation function with depth is taken, although any such function can be used. Formulas obtained are sufficiently general to permit many types of depth-dependent sound speeds and bottom interface models. We illustrate current fluctuation effects on intensity moments for the specific case of an isospeed channel with a perfectly hard bottom. The acoustic consequences of changes in physical parameters directly related to current fluctuations, such as its standard deviation and correlation length, are discussed. Also exhibited are changes in random current effects on intensity which result from changes in noncurrent related parameters, such as acoustic frequency and channel depth. [Work supported by ONR.]

11:00

BBB11. Re-examination of convergence of rough-surface perturbation theory. Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98105)

Several numerical studies have been reported which suggest that the rough-surface perturbation series can give accurate results even when the Rayleigh hypothesis is untrue. The question of convergence or nonconvergence is important for applications of perturbation theory and other theories based upon it (composite roughness, smoothing, phase perturbation). Numerical convergence will be demonstrated for large-amplitude sinusoidal surfaces of sufficiently small slope. Partial theoretical arguments will be given in support of the convergence criterion suggested by the numerical results. [Work supported by ONR.]

11:15

BBB12. Studies of the validity of the Kirchhoff approximation for rough-surface scattering. Eric I. Thorsos (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98105)

The accuracy of the Kirchhoff approximation for randomly rough, pressure-release surfaces has been examined through comparison with exact numerical results based on solving an integral equation. A Gaussian roughness spectrum was chosen for the surfaces, and a Monte Carlo procedure was used to obtain the average bistatic scattering cross section. Results will be presented that illustrate the breakdown of the Kirchhoff approximation as the incident grazing angle is reduced and as the average radius of curvature is reduced. The accuracy of shadowing corrections to the Kirchhoff approximation will also be discussed. [Work supported by ONR.]

11:30

BBB13. Shallow water reverberation: A heuristic ray-mode model. Dale D. Ellis (Defence Research Establishment Atlantic, P.O. Box 1012, Grove Street, Dartmouth, Nova Scotia B2Y 3Z7, Canada)

A shallow water reverberation model has been developed based on normal mode theory together with various ray-mode analogies. The model seems to be different from other shallow water reverberation models [K. V. Mackenzie, J. Acoust. Soc. Am. **34**, 62 (1962); E. C. Shang, 12th ICA Symposium on Underwater Acoustics, Halifax, N.S., Canada (1986)] in that the reverberation at time t is due to contributions from a number of different ranges. Acoustic energy propagates from the source to the scattering element via mode n, and returns via mode m. The scattering is assumed to take place at the water–surface or water–bottom interface in accordance to the scattering function $M(\theta_n, \theta_m)$. The angles are determined from the mode wavenumbers in accordance with the usual ray-mode analogy. The travel times for the different paths are obtained from the modal group velocities. The final result is a double sum over the modes, but is computationally quite efficient since the mode functions and scattering angles are independent of range.

11:45

BBB14. Resonance reflections from a stratified ocean bottom. P. D. Jackins, G. C. Gaunaurd, and J. Arvelo (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20903-5000)

The echoes reflected by a stratified ocean bottom that is insonified by acoustic plane waves emerging from a distant source are studied. The various strata in the bottom are modeled as elastic layers bonded together and allowing longitudinal (acoustic) and transverse (shear) waves to penetrate and be propagated through, and reflected from, the overall configuration. There can be as many layers as one wishes, all of arbitrary compositions. The bottom layer is a half-space of infinite thickness upon which all other layers rest. The top layer is the fluid layer that simulates the ocean and contains the source. The reflections versus frequency at fixed angles of incidence, or versus angle of incidence at fixed frequencies are studied. Numerous resonances are present in these returns and emerge in the analysis. This investigation has the eventual, ultimate, goal of determining details about the bottom composition and stratification, from the remotely sensed reflections. Of particular interest is a five-layer case in which the consolidation and rigidity of the sediments in the various layers increases as one penetrates deeper into the bottom.

Session CCC. Musical Acoustics VIII: Ethnic Musical Instruments, Part 2: Strings

Thomas D. Rossing, Chairman
Department of Physics, Northern Illinois University, DeKalb, Illinois 60015

Invited Papers

2:00

CCC1. Acoustics of the Highland, Irish, and Baroque harps. Ian M. Firth (Department of Physics, University of St. Andrews, St. Andrews, Fife KY16 9SS, Scotland)

Measurements of the string lengths of the extant Highland and Irish harps and the harps of the Baroque period show that there has been a consistent scaling rule used in the construction of the harp from the 14th Century to modern times. Acoustical measurements on extant instruments in museums using portable equipment show that modes of vibration of the soundboards of these instruments can be correlated. Over this period the method of construction of the harp has altered considerably from the Highland and Irish, made of a sound chest hollowed from the solid with string arm and column without any glued joints to the modern harp made of many pieces with a spruce soundboard. The change in construction took place in the Baroque period and measurements show that the modern harp maintains the string scaling rules of the earliest harps of the western seaboard of Europe but incorporates the lighter and more complicated construction of the Spanish type of harp played in the Baroque period. The measured acoustical modes of vibration of a Spanish harp of the Baroque period suggest that the modern concert harp is descended from that type.

2:30

CCC2. Acoustical studies on p'i-p'a and cheng. Shih-yu Feng (Department of Physics, The Chinese University of Hong Kong, Hong Kong)

Improved holographic interference patterns of the vibrational modes of the front plate of a p'i-p'a will be shown. Most of the modes lie below 2000 Hz and those having larger amplitudes of vibration lie between 320 Hz to 1300 Hz. Vibrations of the front plate due to plucking of the strings have been studied. It is found experimentally that any one of the three strongest plate modes can be excited when the fundamental frequency of the plucked string is equal to or is a subharmonic of the frequency of that mode. The p'i-p'a sound generated by plucking the string is picked up by a sound level meter and the Fourier components are obtained by using a digital storage oscilloscope coupled to an Apple II computer. Because of the lack of an anechoic chamber, the sound level meter was placed not more than 1 m from the front plate. Even with this limitation, it was discovered that the sound pressure levels associated with different mode patterns decreased, with increasing distance, at different rates. This seems to mean that listeners at different distances from the player would hear music of different tone quality. Holographic vibration patterns and frequency response curves were also obtained for another Chinese musical instrument, the cheng. Efforts were made to identify various wood or air resonances. The cheng is a much larger instrument than p'i-p'a, and both its front and back plates can vibrate strongly.

Contributed Papers

3:00

CCC3. Vibration properties of the pipa. A Chinese musical instrument. Qing-hua Chen and Jien-ping Yang (Department of Mechanics, Zhejiang University, HangZhou, People's Republic of China)

The pipa is an ancient Chinese musical instrument similar to a lute. Its origin can be traced back 2000 years. It has been referred to as an Oriental guitar. This paper presents the results of theoretical and experimental studies of the vibration properties of a pipa. The spectrum of the sound produced is shown for different playing techniques, playing angles, and playing positions. Sound spectra are related to vibrational modes of the instrument.

3:15

CCC4. Holographic interferometry studies of the Tambura. V. L. Janakiram (Department of Veena, S. V. College of Music and Dance, Tirupati, South India 517 502)

The drone instrument called the Tambura is used in essentially all the classical music concerts in both North and South India. The Tambura has four strings tuned to the notes PSSS, corresponding to G_1CCC_1 of the western system. The length of the Tambura of standard size is usually between 140 and 160 cm. The resonator is of semiglobular shape. The top plank of the resonator is slightly convex and has small holes forming two circles on either side of the bridge. Traditionally Jackwood and Gourds are used for the manufacture of South and North Indian Tamburas, respectively. Under certain conditions, it is found necessary to utilize a suitable substitute wood, such as Silver Oak. Holographic interferometry has been used to examine the modes of vibration of the Tambura's resonator.

Session DDD. Noise IX: Sound Intensity and Sound Power, Part 2

Govindappa Krishnappa, Chairman
National Research Council of Canada, Ottawa, Ontario K1A 0R6, Canada

Chairman's Introduction—1:30

Invited Papers

1:35

DDD1. Advances in the application of sound intensity techniques to the determination of the noise emission of machines. George C. Maling, Jr. (IBM Acoustics Laboratory, Poughkeepsie, NY 12602)

The advances that have been made in the last 50 years in the application of sound intensity techniques to the determination of sound power are reviewed. The early development of sound intensity meters or "acoustic wattmeters" by Olson, Clapp, and Firestone and Schultz are presented, and the development of modern instruments is reviewed. In the early 1970s sound intensity methods were used for the determination of sound power, and improvements in the methods have been made in the last 12 years. In this paper, the effects of background noise and the selection of microphone positions will be discussed, and the progress that has been made in the development of Standards which may be used for the determination of sound power will be presented.

2:05

DDD2. Progress with the ISO and ANSI Standards for the determination of sound power levels of noise sources using sound intensity measurements. Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

Intensity measurements offer the possibility of accurate determination of the *in situ* sound power of machinery noise sources in the presence of high levels of extraneous noise. Meetings of working group ISO/TC43/SCl/WG25 began in 1982. Early meetings were devoted to discussing various measurement difficulties with a draft Standard produced in late 1985. Meetings of working group ANSI S12-21 began in 1983 and produced the first draft of a standard in 1984. It is planned to submit a draft standard to ANSI for trial and study. Although there are many similarities between the ISO and ANSI drafts, there are presently some differences. The ISO draft allows both precision and engineering grade measurements and uses a number of indicators to determine the grade achieved. The ANSI draft presently only allows engineering measurements. The ISO draft concentrates on fixed point measurements on a surface, while the ANSI draft also includes the possibility of scanning.

2:35

DDD3. On the usefulness of active sound intensity for sound power determination. J. Nicolas (G.A.U.S., University of Sherbrooke, Sherbrooke, Quebec JlK 2Rl, Canada)

In order to examine the limitations associated with this technique, a systematic study of the various parameters affecting intensity measurements to determine sound power will be presented. The effect of the finite approximation, phase mismatch, inter-active intensity, the number of points, and the spatial distribution of the points will be discussed, keeping in mind actual measurement conditions. Some *in-situ* experimental studies will be presented. For a typical section of a textile machine, the power measured via ISO 3744 in a semianechoic field compared well with that measured in a semireverberant room using intensity, confirming the validity of the technique. Further, it will be shown that intensity can be used to differentiate the noise emitted by the frame from that of the spindle. A systematic way of including a criterion of precision will be proposed. This leads to a tolerance which gives the degree of confidence on the measurement and which points out clearly unfavorable conditions.

3:05

DDD4. Localization of coherent and incoherent sound sources via an optimization process. M. Sidki and J. Nicolas (Mechanical Engineering Department, Université de Sherbrooke, Sherbrooke, Québec, JlK 2R1, Canada

The problem of the localization and characterization of sound sources is tackled, for the case of a set of monopole sources, starting with the analytic expressions describing the field in combination with a gradient method for the optimization process [M. Box *et al., Nonlinear Optimization Techniques* (Olivier & Boyd, London)]. To locate the sources three measurement variables are used: the 3D vectorial active intensity, the normal active intensity, and the quadratic mean pressure. Simulations for three to five emitting sources, aimed at predicting their positions and spectral densities are made. The minimum number of measurement positions that lead to convergence of the algorithm is investigated. The agreement between prediction and reality is good, even when the sources are coherent. In parallel, the impact of the choice of measurement variable on the precision of the results is studied. In the very near field of the source, it is found that identification via 3D intensimetry requires a significantly lower number of measurement positions than does that via normal intensity or quadratic mean pressure.

3:20

DDD5. A statistical analysis of the estimation error of sound power measured by sound intensity techniques. Mei Q. Wu and Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

The difference between the exact power passing through the measurement surface and the sound power estimated from a finite number of sound intensity measurements is defined as the estimation error. Based on the expression of the error derived in a previous paper [M. Q. Wu and M. J. Crocker, Proc. INTER-NOISE 86, 1129 (1986)], the statistical properties of the error were studied and frequency histograms of it were drawn for the cases where the sound source was assumed to be a point monopole with random locations over the source surface and the measurement surface was a 1-square-meter square parallel to the source surface. It is found that if the measurement distance is 0.3 m and if four sound intensities are measured over the measurement surface, then for 89% confidence the estimation error will be less than 0.9 dB. If the measurement distance is increased to 0.5 m, then for 99% confidence the estimation error will be less than 0.5 dB. Some estimation of the necessary number of sound intensity measurements for a given error limit has also been conducted.

3:35

DDD6. Radiation efficiency as a factor in the design of engine components. Steve Pettyjohn (Peter Klaveness & Associates, 4600 Minnesota Avenue, Fair Oaks, CA 95628)

The importance of radiation efficiency as a factor in the design of engine components is shown. The sound power reduction of a two-piece engine gear cover was the goal. In the past, the action taken was to reduce the vibration amplitude. This ignores the role of radiation efficiency, often assumed to be one. Wallace [C. E. Wallace, J. Acoust. Soc. Am. **51**, 926–945 (1972)] has shown how to analytically calculate radiation efficiency. Three models of the gear cover were developed. The standard gear cover comprises the flat steel plate attached to the block and an aluminum shell covering the gears. The first model, #1, greatly stiffened the shell; the second model, #2, greatly stiffened the flat plate; and the third model, #3, reversed the units, i.e., an aluminum shell was attached to the block and thin steel cover was used to enclose the gears. The model showed decreasing sound power levels from #1 to #3. A prototype part made up to the specifications of number #3 showed sound power measurements, using acoustic intensity methods, in close agreement with predictions.

3:50

DDD7. Using nearfield acoustical holography to analyze a source excited by multiple frequencies. William Y. Strong, Jr. (CBS Technology Center, 227 High Ridge Road, Stamford, CT 06905)

Nearfield acoustical holography (NAH) is a technique that can be used to analyze the radiated sound field located to one side of a source by measuring the pressure amplitude and phase over a surface. Typically a sinusoidal source excitation is used. Additional information could be obtained from a data set if the source was excited at multiple frequencies. These frequencies were each analyzed as if they were obtained from a single frequency measurement. The frequencies would need to be chosen carefully to reduce the influence of one frequency on another caused by spectral leakage when performing a discrete Fourier transform (DFT) on the time data. In the limit, a maximum of $N/2$ analysis frequencies could be measured where N is the number of points in the DFT, if those frequencies were perfectly aligned with the center frequencies of the DFT frequency bins. This presentation will discuss the use of an arbitrary function generator to supply the multiple frequency signal required and will present the results of doing an NAH analysis of a rectangular plate excited by this signal.

4:05

DDD8. Application of nearfield acoustical holography to infrasonic noise source determination. W. S. Gan (Acoustical Services Pte Ltd., 29 Telok Ayer Street, Singapore 0104, Republic of Singapore)

In my previous paper (W. S. Gan, paper presented at the 1st Congresso Brasileiro de Acustica, Brasil, Feb. 1979), we apply acoustical holography to the determination of infrasonic noise source with the experimental setup using a correlation filtering approach. In this paper, we apply acoustical holography to the same problem, but with the experimental setup using the nearfield acoustical holography (NAH) [E. G. Williams and J. D. Maynard, Phys. Rev. Lett. **45**, 554–557 (1980)] which can overcome the resolution problem. Infrasonic frequencies are even more appropriate for NAH than are audio frequencies due to the very long wavelengths involved. We propose the use of array processing since the experimental setup will involve large number of microphones (sensors). Array processing will speed up the calculation of FFT and Green's function. It permits the simultaneous processing of the data from each sensor. We derive the Green's function and the expression for the holographic reconstruction from the Bojarski's exact inverse scattering theory [N. N. Bojarski, Radio Sci. **16**, 1025 (1981)]. An exact closed-form analytic solution for the infrasonic noise source term is obtained.

Session EEE. Physical Acoustics VIII: Sound Propagation and Attenuation

Frederick H. Fisher, Chairman
Marine Physical Laboratory, Scripps Institute of Oceanography, University of California, La Jolla, California 92093

Contributed Papers

1:30

EEE1. Effect of laser pulse energy on optoacoustic conversion in liquids. Henry E. Bass and Stanley A. Cheyne (Physical Acoustics Research Group, The University of Mississippi, University, MS 38677)

The effect of laser pulse energy on optoacoustic conversion has been studied experimentally. A short UV laser pulse (337 nm in wavelength and 800 ps in duration) is allowed to impinge on a test cell of propanol where it is absorbed in a short distance ($\sim$0.87 mm). The absorbed energy results in a density variation in the fluid which propagates outward from the excitation zone. A second, 10-μm-diam cw He–Ne probe laser beam parallel to the excitation pulse is deflected by the passing wave. The deflection is measured with a photodiode [B. Sullivan and A. C. Tam, "Profile of laser-produced acoustic pulse in a liquid," J. Acoust. Soc. Am. **75**, 437–441 (1984)]. The time evolution as well as the amplitude of the measured signal is considered. The amplitude of the optoacoustic signal is found to vary linearly with input energy between 7.5×10^{16} W/m^3 and 5.3×10^{15} W/m^3. Theoretical limits on the observed linearity will be presented. [Work supported by the ONR.]

1:45

EEE2. Sound amplification from controlled excitation reactions (SACER) in N_2/He and N_2/CH_4 mixtures. F. Douglas Shields (Department of Physics and Astronomy, University of Mississippi, University, MS 38677) and L. Dwynn Lafleur (Department of Physics, University of Southwestern Louisiana, Lafayette, LA 70504)

Last year, the discovery of SACER in N_2/H_2 mixtures was reported [J. Acoust. Soc. Am. Suppl. 1 **78**, S45 (1985)]. This paper discribes similar measurements in N_2/He and N_2/CH_4 mixtures. Sound amplification has been observed, and its dependence upon gas pressure, sound frequency, foreign gas concentration, and vibrational temperature is that generally expected from a crude relaxation model for the gases and published values of the relaxation times. However, as in the case with N_2/H_2 mixtures, the gain is an order of magnitude bigger and persists longer than expected. An electrical discharge was passed through the gas to initiate the acoustical oscillations and elevate the vibrational temperature. In N_2/CH_4 mixtures, the gain and the nature of the electrical discharge were found to depend on the number of times the discharge was passed through the gas. This result is interpreted as being due to a chemical reaction between N_2 and CH_4 that produces an impurity that shortened the relaxation time. [Work sponsored by the ONR.]

2:00

EEE3. Calculated *B/A* parameters for *n*-alkane liquids. Bruce Hartmann and Edward Balizer (Polymer Physics Group, Naval Surface Weapons Center, Silver Spring, MD 20903-5000)

The B/A parameters for a series of n-alkane liquids of chain lengths $n = 5,7,8,9,12,16$, and ∞ (i.e., molten polyethylene) were calculated from an analytic PVT equation of state, supplemented with experimental heat capacity data. The density, thermal expansion coefficient, sound speed, and sound-speed derivatives (with respect to temperature and pressure) all vary in a systematic manner with chain length and approach the polyethylene values asymptotically. The pressure derivative of the sound speed is calculated much more accurately than the temperature derivative; but since the pressure derivative term dominates, the overall accuracy of the calculations is close to the experimental accuracy. Calculated density values agree with experiment to within 0.2% while B/A values, which require a second derivative of the density, are within 5% of the values calculated numerically from experimental sound speeds in the literature.

2:15

EEE4. Reflection characteristics for an immersed transversely isotropic layer. Aleksander Pilarski and Joseph L. Rose (Department of Mechanical Engineering and Mechanics, Drexel University, Philadelphia, PA 19104)

In order to obtain more information about a composite material with respect to material characterization and defect analysis, utilizing oblique ultrasonic beams or guided waves inside the structure is proposed. Measurements of ultrasonic reflectivity from composite plates immersed in a liquid have been carried out. Initial experiments have been performed on a undirectional graphite–epoxy composite. To choose the appropriate testing parameters (transducer frequency, angle of incidence, etc.) and to provide a quantitative analysis of the obtained results, model calculations were carried out for a transversely isotropic fluid-loaded layer using Thompson's recurrence formulas connecting the wave amplitudes in neighboring media. Employing long wavelengths, it was assumed that the layer was homogeneous with five effective, independent elastic coefficients. The calculations were carried out for different values of volume fractions of the voids and fibers. These data have been compared to experimental results. The experimental measurements have been performed using frequency spectrum analysis of a pulse which has passed through a composite layer in the fiber direction. Experimental and theoretical results are encouraging. [Work supported by NRL.]

2:30

EEE5. Split mode excitation of traveling waves in ring resonators. Peter H. Ceperley (Department of Electronic and Computer Engineering, George Mason University, Fairfax, VA 22030)

Traveling wave excitation in ring resonators has application to surface wave motors [H. Yamamoto *et al.*, U.S. Patent 4,504,760 (1985)] and thermoacoustic heat engines [P. Ceperley, J. Acoust. Soc. Am. **77**, 1239–1244 (1985)] as well as in other potential areas. The standard method of exciting traveling waves in ring resonators involves two wave sources spaced a quarter-wavelength apart and driven with a 90° phase difference. This paper presents a method of introducing a perturbation into the ring resonator that allows excitation of traveling waves with a single exciting source. This method also has application to similar rotating mode excitation in simply connected resonators that approximate objects of revolution.

2:45

EEE6. Leaky Rayleigh wave reflection from anisotropic wedges. Michel de Billy,[a] Peter B. Nagy, Gerard Quentin,[a] and Laszlo Adler (Department of Welding Engineering, The Ohio State University, 190 West 19th Avenue, Columbus, OH 43210)

In order to study the interaction of surface waves at anisotropic boundaries, a single crystal wedge was used in an immersion broadband experiment. A new method was introduced which separates the reflection coefficient due to anisotropy from geometrical effects. According to this approach, the reflection coefficients of the surface wave due to the geometry of the sample are the same from both directions, while the anisotropic component of the reflection coefficients are of the same modulus but opposite in phase. The leaky Rayleigh wave attenuation coefficient's frequency dependence was also measured, and accounted for the reflection coefficient. [Work partially supported by DOE and NATO.] [a] Permanent address: Universite Paris 7, 2 place Jussieu, Paris Cedex 05, France.

3:00

EEE7. An experimental study of the acoustical impedance and fluid mechanics of two orifices in series. Donald F. Elger and Ronald L. Adams (Tektronix, Inc., P.O. Box 500, M.S. 50-321, Beaverton, OR 97077)

Certain ink-jet printer nozzles consist of two orifices in series. Since the pressure impinging on the nozzle is transient, a study of nozzle impedance and fluid mechanics was motivated. Air was used, and the orifices were fabricated in equal diameter plates attached to a piston driven cylinder. The space between the plates, and the outer orifice, were open to ambient. The independent variables studied were: $0.23 \leqslant d_2/d_1 \leqslant 0.94$; $0.4 \leqslant t/d_1 \leqslant 2.6$; and $660 \leqslant P'/\rho\omega v \leqslant 4120$, where d_1 and d_2 are the inner and outer orifice diameters, ω is frequency, P' is the acoustic pressure magnitude, and t is plate spacing. Hot-wire anemometry and a standing wave resonance method were used to obtain nozzle impedance and air velocity measurements. The impedance data showed the nonlinear behavior characteristic of a single orifice. The velocity data showed that t/d_1 has the largest effect on the magnitude and nonlinearity of the fluid velocity at the outer orifice.

3:15

EEE8. Vibrational–vibrational coupling in air at low humidities. Allen J. Zuckerwar (NASA Langley Research Center, MS 238, Hampton, VA 23665) and Keith W. Miller (Virginia Commonwealth University, Department of Mathematical Sciences, Richmond, VA 23284)

Calculations of sound absorption in air are traditionally based on the assumption that molecular relaxations in nitrogen and oxygen are independent. In binary mixtures of these two gases, however, molecular relaxation is known to be controlled by a very strong vibrational–vibrational (V–V) coupling, which influences both the relaxation times and the relaxation strengths. This presentation addresses the question of whether small concentrations of the air constituents carbon dioxide and water vapor, which themselves possess a strong V–V coupling to N_2 and O_2, serve to decouple the N_2 and O_2 relaxations. A "coupling strength" is derived, which depends upon the constituent concentrations and the related reaction rate constants. It is found that the molecular relaxations associated with N_2 and O_2 in air experience a gradual transition from strong to weak coupling as the humidity increases beyond approximately one part per million.

3:30

EEE9. A transfer function approach to transient propagation in media with a quadratic frequency dependence of attenuation coefficient. Daniel Guyomar (Schlumberger-EPS, 26 rue de la Cavee, 92140 Clamart, France) and John Powers (Department of Electrical and Computer Engineering, Naval Postgraduate School, Monterey, CA 93943)

The propagation of short acoustical pulses in lossy media is presented as a time-varying transfer function (or spatial filter) that acts on the spatial spectrum of the source excitation. The source is a planar source of arbitrary (but separable) spatial and temporal dependence mounted in a rigid baffle. The homogeneous medium obeys Stokes' equation, resulting in a quadratic dependence of the plane-wave attenuation coefficient. Using distribution theory, an integral expression for the field is derived in terms of the normal velocity of the source, the Green's function, and its normal derivative on the boundary. This expression leads to an approximate solution for a point source in the source plane, which in turn produces a spatial transform that is the propagation transfer function. This transfer function is simply related to the transfer function of lossless propagation. Computer simulations of fields as well as a discussion of some casuality problems introduced by the use of the Stokes' equation are presented.

3:45

EEE10. Comparison of constrained layer damping models. Donald W. Fausett, Laurene V. Fausett (Department of Mathematical Sciences, Florida Institute of Technology, 150 West University Boulevard, Melbourne, FL 32901), and Pieter S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

The constrained layer damping technique is based on the application of a layer of viscoelastic material onto the surface to be damped and beneath a layer of stiff material. The damping mechanism is dissipation of energy caused by shear motion in the damping layer. An early model for the propagation of straight-crested waves in a constrained plate by Kerwin [J. Acoust. Soc. Am. **31**, 952–962 (1959)] uses thin-plate theory for the base plate and is limited to flexural waves. This model was extended to cover extensional as well as flexural waves in thick plates by describing the base plate by exact elasticity theory [P. S. Dubbelday, J. Acoust. Soc. Am. **80**, 1097–1102 (1986)]. The present study proposes the formulation of a model based on exact elasticity theory for all three layers. A computer program based on the exact equations for the case of an infinite composite plate in vacuum has been developed. It is used as a basis for comparison of results from previous models and to investigate certain anomalies in the calculated results of those models. [Work supported by the ONR.]

4:00

EEE11. Viscoelastic versus radiation damping in fluid-loaded constrained plates. Pieter S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337)

A hybrid model was developed for the analysis of constrained-layer damping whereby the base plate is described by exact elasticity theory, while the elastomer and constraining layers are analogous to Kerwin's model [J. Acoust. Soc. Am. **31**, 952–962 (1959)]. Under fluid loading, the success of the technique is measured by a damping efficiency, the ratio of loss in the elastomer to the total loss due to viscoelastic and radiation damping. This efficiency is usually evaluated by combining the viscoelastic loss from a composite plate in vacuum, for antisymmetric and symmetric waves separately with the radiation loss from a single fluid-loaded plate. This assumes simple addition of the two effects for a fluid-loaded constrained plate. Extension of the hybrid model to include fluid loading shows that at higher frequencies the presence of the two types of waves redistributes the energy across the plate, invalidating the presumed additive property. Thus it proves necessary to compute damping efficiency from analysis of the fluid-loaded composite plate with inclusion of both antisymmetric and symmetric waves. [Work supported by the ONR.]

4:15

EEE12. Damping of acoustic waves in bcc crystals. Kailash and Kiran Shanker (Department of Physics, University of Allahabad, Allahabad 211002, India)

The study of damping of acoustic waves is made in bcc crystals from 50–500 K at zero magnetic field. The theory is developed from primary physical constants; nearest-neighbor distance and hardness parameter; and basic potentials: electrostatic and Born–Mayer. The cesium halides are studied acoustically along different directions of propagation and orientation. The molecular weight, temperature, frequency, direction of propagation, and orientation dependence are explained, and it is stated that damping is the characteristic property of substance.

4:30

EEE13. Damping of elastic waves in metallic alloys. Kiran Shanker and Kailash (Department of Physics, University of Allahabad, Allahabad 211002, India)

Starting from the nearest-neighbor distance and repulsive parameter and using Coulomb and Born–Mayer potentials, the theory is developed for damping of elastic waves in metallic alloys. The study is made in nickel–copper alloys for different compositions over a wide temperature span. It is observed that the damping follows the T^n law, where n is an exponent. It is concluded that damping is the fundamental property of the substance.

4:45

EEE14. Dynamic elastic moduli of several thermoplastic elastomers for use in acoustic systems. Anthony B. Bruno and Rolf G. Kasper (Research and Technology Staff, Naval Underwater Systems Center, New London, CT 06320)

A number of dynamic mechanical measurements have been performed on the Metravib-04 viscoanalyzer for a variety of thermoplastics. The material was a thermoplastic elastomer composed of styrene ethylene butylene styrene (SEBS) as a copolymer base, plasticized with a parfinic oil, and blended with a high-density polyethylene plus a stabilizer. The tested materials differed by the amount of high-density polyethylene blended into each sample. Each material was tested in tension compression and in shear for the frequency range 5 Hz–1 kHz at room temperature. The data yielded independent estimates of complex elastic moduli, e.g., Young's modulus and shear modulus as a function of frequency. The data showed good correlation between elastic moduli and percentage of polyethylene in each sample. However, no corresponding correlation with damping was observed. The dilatational and shear wave speeds and the Poisson's ratio were also computed as a function of frequency for each sample. These data were used to evaluate the performance of various acoustic systems. A number of examples will be discussed.

FRIDAY AFTERNOON, 12 DECEMBER 1986 — SALON F, 1:30 TO 3:45 P.M.

Session FFF. Psychological and Physiological Acoustics VIII: Auditory Adaptation and Sensitivity

Bertram Scharf, Chairman
Auditory Perception Laboratory, Northeastern University, Boston, Massachusetts 02115

Contributed Papers

1:30

FFF1. Annoyance of moderate level of noise spectra with predominating pure tones. Gordon R. Bienvenue, Brian Wood, Susan R. Glass (Department of Communication, State University of New York, College at New Paltz, New Paltz, NY 12561), and Robert D. Celmer (Department of Engineering, University of Hartford, West Hartford, CT 06117)

Annoyance due to noise has been a concern to researchers for over 30 years. Recently, concern has centered upon the annoyance engendered by noise spectra with predominating pure tones since the effect of tones on annoyance is marked and because modern business equipment often demonstrates this type of spectrum. Though most business equipment operates at sound levels at or below 65 dB SPL, much of the annoyance research has been directed towards aircraft noise and has been conducted with noise spectra exceeding 80 dB SPL. The present research centers on the overall levels of primary concern in evaluating business equipment annoyance while maintaining a link to the established (high level) data base. Twenty-five subjects were presented with 1200 tape recorded sounds of varying tone-noise ensembles including some business machinery sounds. The subjects rated these sounds for their levels of loudness, tonal prominence, and annoyance. Results of this procedure are discussed with implications for noise control. Of particular interest was the effect of modifying the tonal components of real equipment spectra in both amplitude and frequency using a digital signal processing system. [Work supported by IBM Corporation.]

1:45

FFF2. The effect of combinations of continuous and impact noise on the auditory system: Energy considerations. William A. Ahroon, Roger P. Hamernik (Auditory Research Laboratories, SUNY, Plattsburgh, NY 12901), and Richard J. Salvi (Callier Center for Communication Disorders, University of Texas, Dallas, TX 75235)

Fifty-three chinchillas completed an experimental paradigm designed to study the interaction between a low-frequency octave band of noise and three levels of impact noise whose parameters were balanced to produce exposures having approximately equal energies. The animals were exposed to one of the following seven conditions: (1) 0.5 kHz C.F. octave band @ 95 dB SPL for 5 days; (2) 113 dB peak SPL impact noise, 1/s for 5 days; (3) Combination of 1&2; (4) 119 dB peak SPL impact noise, 1/4 s for 5 days; (5) Combination of 1&4; (6) 125 dB peak SPL impact noise, 1/16 s for 5 days; (7) Combination of 1&6. Pre- and post-exposure estimates of threshold were obtained using evoked auditory potentials recorded from a chronic electrode implanted in the area of the inferior colliculus. Status of the sensory cell population was evaluated using surface preparation histology. The three energy balanced "impulse-alone" exposure conditions produced very similar levels of ATS, PTS, and sensory cell loss indicating that the exposure energy is a critical variable for predicting the consequences of such exposures. However, the three combination exposures produced results which differed across the three groups with the 119- and 125-dB combination conditions producing a significantly increased level of acoustic trauma. The results of the combination exposures, cast doubt upon the general validity of the equal energy concept. [Work supported by NIOSH.]

2:00

FFF3. Relationship between extra-high-frequency hearing and cardiovascular endurance in noise-exposed industrial workers. Amy Donahue (U. S. Army Environmental Hygiene Agency, Bio-Acoustics Division, Aberdeen Proving Ground, MD 21010-5422) and Allan O. Diefendorf (Department of Audiology, University of Tennessee, Knoxville, TN 37916-0740)

Extra-high-frequency audiometry may have a clinical use as an indicator of the traumatic effects of noise. The purpose of this study was to determine if cardiovascular endurance has a relationship to the extra-high-frequency thresholds of hearing. Industrial workers (ages 27–30, with normal hearing in the conventional frequency region) exposed to daily noise doses of 85–88 dBA served as subjects. Cardiovascular endurance and extra-high-frequency thresholds were determined for 50 subjects. A two-factor ANOVA (Group × Frequency) revealed no significant differences in the extra-high-frequency thresholds of two groups of industrial workers with varying cardiovascular endurance. For 50 subjects, no significant correlations existed between the thresholds of hearing at the extra-high frequencies and cardiovascular endurance. It was concluded that cardiovascular endurance cannot be used to determine individual susceptability to the noxious effects of noise.

2:15

FFF4. Temporary auditory-threshold shifts induced by exposures to continuous tones in water. Paul F. Smith (Naval Submarine Medical Research Laboratory, Groton, CT 06349)

Four bare-headed divers were exposed for 25 min to continuous tones in water at frequencies of either 700, 1400, or 5600 Hz at average sound pressure levels (SPL) between 143.1 and 165.1 dB above 20 μPa. These SPL are comparable to those produced by some underwater hand-held tools. These conditions yielded individual temporary auditory-threshold shifts (TTS) between 23 and 55 dB 2 min post-exposure. Recovery required up to 50 h or more depending on exposure conditions. Nonauditory effects including middle-ear sensations and a reddened ear drum were also observed. Four additional divers were exposed successively for 10-min periods to pure tones of 5600, 1400, and 700 Hz (in that order) in air at 100 dB and in water at various SPL between 125 and 150 dB. Divers incurred moderate TTS from all exposures. For equivalent intensities in air and in water, 700-Hz exposures produced equivalent TTS. Less TTS was induced in water at 1400 Hz; more TTS in water at 5600 Hz. The results suggest that, within the predictive accuracy of the TTS technique, several hand-held tools now in use by military and civilian divers are extremely hazardous to hearing.

2:30

FFF5. Mathematical model of the effect of limited stapes displacement on hazard from intense sounds. G. Richard Price and Joel T. Kalb (U. S. Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005-5001)

As a follow-on to an idea presented earlier [G. R. Price, J. Acoust. Soc. Am. **56**, 195–197 (1974)] an integrated mathematical model of the external, middle, and inner ears has been developed for the cat ear. It allows the calculation of stapes displacements (among other values) in response to any free-field sound pressure. An absolute limit to displacement of about 20 μ peak to peak is assumed, based on the dimensions of the annular ligament and in accord with the measurements of Guinan and Peake [J. Acoust Soc. Am. **41**, 1237–1261 (1967)]. Where hazard from intense impulses, such as gunfire, are concerned, the effect of this limitation on stapes displacement is profound, producing a spectral bias (higher velocities being transmitted at high frequencies) as well as the blocking of transmission into the cochlea when amplitudes exceed the limiting values. This action may in part explain why low-frequency impulses are much less hazardous than their energy content would seem to indicate as well as why hearing losses are not more severe for very intense sounds (150 dB and up) commonly found around weapons.

2:45

FFF6. Additivity of induced loudness adaptation. Bertram Scharf, Georges Canévet,[a] and Søren Buus (Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

The loudness of a mid-level tone can be made to go down in at least two ways: (1) by intermittently incrementing the tone, (2) by presenting an intermittent tone to the contralateral ear. Magnitude estimates given during the intervals when the ipsilateral increment or contralateral tone is absent have shown that loudness decreases approximately 50% in 2–3 min. The present experiments investigated the effect of combining the ipsilateral and contralateral intermittencies. Observers judged the loudness of a monaural steady tone (0.25, 0.5, 1, or 4 kHz at 50 dB SPL) presented: (a) alone, (b) with a 20-dB increment every 20 s for 10 s, (c) with a 50-dB tone to the contralateral ear every 20 s for 10 s, (d) under both (b) and (c). Under (a), loudness changed little over 2–3 min. Under (b), loudness decreased about 60%. Under (c), loudness decreased by 50% at 4 kHz, but less at lower frequencies. (This frequency effect occurs because contralateral intermittency appears to induce adaptation in fewer subjects at low than at high frequencies.) Under (d), loudness decreased less than predicted for independent summation of ipsilaterally and contralaterally induced loudness adaptation. [Work supported by NIH-NINCDS grant ROI-NS07270]. [a] Current address: CNRS, LMA, Marseille, France.

3:00

FFF7. Pure-tone sensitivity of human infants. Lynne Werner Olsho, Elizabeth G. Koch, Elizabeth A. Carter, Christopher F. Halpin, and Nancy B. Spetner (Department of Otolaryngology, RL-30, University of Washington, Seattle, WA 98112)

The ability of 3-, 6-, and 12-month-old infants to detect pure tones in quiet was tested at frequencies ranging from 250–8000 Hz. Stimuli were presented monaurally via headphone. Signal trials consisted of ten repetitions of a 500-ms tone burst, with 10-ms rise–fall time and 500 ms between bursts; no-signal trials were 10-s intervals of quiet. The infant's response to a tone was judged by an observer, who, blind to trial type, decided whether or not a tone had been presented on each trial, based on the infant's behavior. Comparison of infant thresholds, determined using an adaptive rule, to those of adults tested under similar conditions showed a progressive improvement in threshold from 3–12 months. Three-month-olds' threshold were relatively poorer at 250 and at 8000 Hz compared to adults. The 6- and 12-month-olds' thresholds were somewhat closer to those of adults at 4000 and at 8000 Hz than at lower frequencies. Maturation of absolute sensitivity, of frequency resolution, and of nonsensory processing may all contribute to these age-related changes. [Work supported by NIH.]

3:15

FFF8. A comparison of two-, three-, and four-alternative forced-choice staircase procedures. R. S. Schlauch (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195) and R. M. Rose (Department of Psychology, University of Washington, Seattle, WA 98195)

Threshold estimates from multiple-interval forced-choice staircase procedures were studied using computer simulations. Two frequently cited psychometric function shapes, those for the signal known exactly and for high uncertainty, governed the hypothetical subject's responses in the simulations. Threshold estimates based on 100 trials were obtained for both psychometric functions, for three different stimulus change values, and for decision rules that estimated 70.7% and 79.4% correct levels. A total of 1000 replications in each condition were run. Each threshold estimate was calculated by averaging the stimulus levels at which a reversal in stimulus direction change occurred. For many of the simulated conditions, three-alternative forced choice was the most efficient procedure (i.e., the variability of the estimates per unit of collection time was smallest). The validity of the simulations was tested using human performance data. [Work supported by NINCDS and IBM.]

3:30

FFF9. Ascending and descending thresholds of pure tones. I. M. Young and L. D. Lowry (Department of Otolaryngology, Jefferson Medical College of Thomas Jefferson University, Philadelphia, PA 19107)

pairment. The ratio of 2:1 was the same as the speed of mechanical attenuation rates. The ratio of 1.6–1.7:1 may suggest that the slower attenuation rate decreased speed of machine relatively slower than mechanism of adaptation. In conventional automatic audiometry, pulsed tones are interrupted 2½ times per second with an off-time of 200 ms. For this stimulus, substantial amount of adaptation can be demonstrable even in normal ears if threshold is approached from above-threshold stimulus levels. The results were compared and discussed with our previous data in that an off-time of 700 ms or longer is required for complete recovery from adaptation when descending method is used.

Fixed frequency Békésy threshold measurements were made for subjects with normal hearing and hearing impairment. Ascending (no sound to sound) and descending (sound to no sound) methods were used for both continuous tones and tones interrupted 2½ times per second with attenuation rates of 2.5 and 5.0 dB/s. For ascending method, the ratio of time to reach the first appearance of tone by slow and fast attenuation rates was 2:1. For descending method, the ratio of time to reach the first disappearance of tone was 1.6–1.7:1. There were no observable differences in these ratios for various frequencies, starting intensities, interrupted and continuous tones, and for subjects with normal hearing and hearing im-

FRIDAY AFTERNOON, 12 DECEMBER 1986 SALON E, 1:30 TO 4:30 P.M.

Session GGG. Speech Communication VIII: Speech Perception 2

Carolyn Wardrip-Fruin, Chairman

Department of Communicative Disorders, California State University, 1250 Bellflower Boulevard, Long Beach, California 90840

Contributed Papers

1:30

GGG1. Perceptual basis for the compact-diffuse distinction for consonants. Sarah Hawkins (Department of Linguistics, University of Cambridge, Cambridge CB3 9DA, England) and Kenneth N. Stevens (Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

A distinguishing acoustic property of velar obstruents is a prominent midfrequency "compact" spectral peak (roughly 800–2000 Hz). Is there evidence for a qualitatively different auditory response for sounds with compact as opposed to diffuse spectra? We generated two ten-item series of bursts by exciting a group of resonators with white noise and systematically varying the relative amplitudes of the spectral peaks corresponding to formants 2 to 5. Stimuli at one end of each continuum had a compact, *k*-like spectrum and at the other end a diffuse *p*-like or *t*-like spectrum. The bursts were appended to short, transitionless vowels with or without intervening aspiration. Identification functions were obtained for the syllable-like stimuli (with response alternatives p t k b d g) and for the isolated bursts (using an ABX paradigm). The results suggest that a sound with a compact spectral peak can cause a distinctive perceptual response when that peak is sufficiently narrow and projects well above adjacent spectral minima. Responses are also influenced by the amplitude of the burst in relation to the following vowel and by the burst duration. Possible mechanisms underlying this distinctive response are discussed. [Work supported in part by NINCDS.]

1:42

GGG2. Basic sensitivity and context memory on two-consonant continua. Rina Goldberg (Research Laboratory of Electronics, MIT, Cambridge, MA 02139 and AT&T Bell Laboratories, Naperville, IL), Neil A. Macmillan (Department of Psychology, Brooklyn College, Brooklyn, NY 11210 and Research Laboratory of Electronics, MIT, Cambridge, MA 02139), and Louis D. Braida (Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

According to the Perceptual Anchor Model of Context Coding [Braida *et al.*, J. Acoust. Soc. Am. **76**, 722–731 (1984)], resolution tasks differ in memory coding requirements: basic sensory limitations are measured in fixed discrimination, while context memory limitations can be inferred by comparing fixed discrimination with identification. The theory was applied to two synthetic consonant continua, one varying in voicing (from /ba/ to /pa/), the other in place of articulation (/ba/–/da/–/ga/). For both continua, regions of high basic sensitivity were found between the best phonemic exemplars, while optimal context coding occurred near the exemplars. Neither result occurs for an /i/–/ɪ/–/ɛ/ vowel continuum [Goldberg *et al.*, J. Acoust. Soc. Am. Suppl. 1 **77**, S7 (1985)]. [Work supported by NSF.]

1:54

GGG3. Amplitude versus frequency modulation in speech perception. H. T. Bunnell and J. M. Pickett (Sensory Communication Research Laboratory, Gallaudet University, Washington, DC 20002)

Remez *et al.* [Science **212**, 947–950 (1980)] presented stimuli consisting of three tones that were simultaneously frequency and amplitude modulated to follow variations in the frequencies and amplitudes of the first three formants throughout an utterance. Bunnell [J. Acoust. Soc. Am. Suppl. 1 **75**, S85 (1984)] presented speech consisting of five amplitude modulated tones at fixed frequencies 1000 Hz apart. Both types of stimuli represent a severe reduction of speech information, however, one might expect that stimuli like those used by Remez *et al.*, because they more clearly represent formant frequency variations over time, would provide better information for phonetic perception. To directly compare these two stimulus types, nonsense utterances of the form /əCalə/ (C = {/b/,/d/,/g/}) were synthesized using natural productions as templates. In one condition (FM), tokens consisted of four tones which followed the frequencies of the first four formants. In the other condition (AM), stimuli consisted of tones at 500, 1500, 2500, and 3500 Hz which were independently modulated in amplitude to follow variations in the utterance's spectrum level at those frequencies. The stimuli were presented to listeners in a three-alternative stop consonant identification task both with and without added broadband noise. Results suggest that, with some constraints, AM tokens supported stop consonant identification at least as well as their FM counterparts.

2:06

GGG4. Importance of various frequencies to perception of continuous discourse. Chaslav V. Pavlovic (Speech and Hearing Center, University of Iowa, Iowa City, IA 52242), Gerald A. Studebaker, and Robert L. Sherbecoe (Memphis Speech and Hearing Center, 807 Jefferson Avenue, Memphis, TN 38105)

Normal-hearing subjects estimated intelligibility of continous discourse passages spoken by three talkers under various conditions of filtering and noise interference. The relationship between the intelligibility of continuous discourse and the articulation index (transfer function) was different from any found in ANSI S3.5-1969. Also, the lower frequencies were found to be relatively more important for the intelligibility of continuous discourse than for identification of nonsense syllables and other types of speech for which data are available except for synthetic sentences.

2:18

GGG5. The influence of spectral and temporal properties on stop-consonant perception for two-formant stimuli. Ralph N. Ohde (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232)

The results from a previous study on stop-consonant perception for two-formant stimuli showed that the most reliable responses were for those stimuli with substantial second-formant ($F2$) transitions, and that there were few consistent velar responses [R. N. Ohde and K. N. Stevens, J. Acoust. Soc. Am. Suppl. 1 **75**, S66 (1984)]. From these results, it was hypothesized that consistent velar responses may only be obtained by delaying the frequency increase in the first formant ($F1$) or by making the $F1$ transition slower. The aim of this study was to test this hypothesis by systematically varying the duration of $F1$ between 10 and 40 ms, and the duration of an onset of a single pulse preceding formant frequency excitation between 10 and 20 ms. Two two-formant CV stimulus continua were synthesized, each covering a range of $F2$ starting frequencies, for vowels corresponding roughly to [ɪ, ɑ]. Subjects identified these syllables under both an open and a forced-choice response set. The results showed that as the duration of these temporal properties increased, velar responses increased for some stimuli. However, considerable variability was observed across subjects. The results will be discussed relative to trading relations between spectral and temporal properties, and relative to their practical and theoretical implications. [Work supported in part by a Vanderbilt University Research Council Grant.]

2:30

GGG6. An auditory basis for rate normalization in stops and glides. Randy L. Diehl and Margaret A. Walsh (Department of Psychology, University of Texas, Austin, TX 78712)

When cued by formant transition duration, the perceptual distinction between stops and glides (e.g., [b] vs [w]) is subject to a "rate normalization" effect: longer steady-state vowel segments shift the perceptual boundary toward longer transition durations [J. L. Miller and A. M. Liberman, Percept. Psychophys. **25**, 457–465 (1979)]. However, rate normalization apparently does not occur when the stop–glide distinction is primarily signaled by rise time [P. C. Shinn, S. E. Blumstein, and A. Jongman, Percept. Psychophys. **38**, 397–407 (1985)]. Both sets of findings have essentially been duplicated using nonspeech analog stimuli. In one series of experiments, the stimuli consisted of a frequency-ramped tone of varying duration, immediately followed by either a short or long steady-state sinusoid. The effect of the long steady-state segment was to shift the categorization boundary toward longer ramp durations. In another series of experiments, the stimuli consisted of a fixed-frequency sinusoid or square wave with varying rise time and either a short or long peak amplitude interval. Categorization on the basis of rise time was little affected by peak amplitude duration. Given these findings, it is suggested that the earlier speech results have a basis in general audition. [Work supported by NICHD.]

2:42

GGG7. Auditory confusion of whispered plosives. Hsiao-Chuan Chen and Igor V. Nábělek (Department of Audiology and Speech Pathology, College of Liberal Arts, University of Tennesse, Knoxville, TN 37996-0740)

Ten normal-hearing subjects listened to a recorded list of 120 whispered consonant-vowel syllables spoken by a male talker. The consonants were six plosives /b,p,d,t,g,k/, the vowel was /a/. Each syllable was repeated 20 times in random order. The overall consonant identification score was 85.2%. The scores for the voiced consonants were lower than for the voiceless ones, but they were still rather high. The highest average identification score of 96% was obtained for /k/. The average scores were 91%, 88.5%, 83%, 81%, and 71.5% for /t/, /d/, /p/, /b/, and /g/, respectively. A pairwise comparison showed that the difference between /k/ and /g/ scores was significant. With respect to place, alveolars and bilabials had the highest and the lowest scores, respectively. Whispered /b,d,g/ were more often confused with their cognates than vice versa; cognate errors were more numerous than place errors. These results indicate that the voicing contrast is indeed useful for distinguishing voiced and voiceless plosives, however that there are other cues which allow the discrimination between voiced and voiceless consonants if the voicing contrast is absent.

2:54

GGG8. Analysis and perception of sibilant fricatives: Shona data. Christopher Clark, Anthony Bladon (Phonetics Laboratory, University of Oxford, 37-41 Wellington Square, Oxford, OX1 2JF, United Kingdom) and Katrina Mickey (School of Oriental and African Studies, University of London, Malet Street, London WC1E 7HT, United Kingdom)

In addition to the commonly found [s,z] and [ʃ,ʒ], the Shona language has a third pair of sibilants $\left[\overset{w}{s},\overset{w}{z}\right]$ usually described as labialized alveolar fricatives and sometimes called "whistling" fricatives. Analysis of all these sibilants revealed that the location of the main frequency peak was typically 3 Bark higher for [s,z] than for the remaining four fricatives which did not differ significantly in this respect. Durational differences were slight throughout, and seemed unlikely to contribute as place cues. A likelier possibility for cueing the place distinction, we hypothesized, was an interaction of peak location with the gradient of the high- and low-frequency slopes adjacent to the peak, together perhaps with contributions from the VCV transitions. Experimental manipulation of these variables in synthetic versions of the three voiceless sibilants yielded listener identifications suggesting that (1) a flat HF slope cued [s] overridingly, irrespective of the peak location, but (2) peak location became important when a falling HF slope is used; and (3) a steep LF slope failed to create enough of a "whistling" percept as to outweigh transitional cues to the [ʃ]-$\left[\overset{w}{s}\right]$ distinction.

3:06

GGG9. Dialect differences and the perception of the fricative–affricate–stop distinction in English. Carolyn Wardrip-Fruin (Department of Communicative Disorders, California State University, Long Beach, CA 90840)

The fricated portions of the naturally produced words *ship, shop, shoe* (as produced by a standard English speaker) were systematically reduced by computer editing. These edited syllables were randomized and presented to 20 adult listeners, 10 of whom spoke a variety of standard English and 10 of whom spoke a variety of nonstandard English which has been influenced by Spanish. Listeners judged each syllable as having initial fricative, affricate, or stop (e.g., ship, chip, or tip). Results show a crossover in judgments from fricative to affricate (e.g., ship to chip) and from affricate to stop (e.g., chip to tip) in specific ranges of fricative durations for both groups of listeners. However, the range of durations judged ambiguous was greater for the speakers of the Spanish influenced dialect particularly between the fricative and affricate categories. This may be a reflection of the phonological status of this distinction in the nonstandard dialect.

3:18

GGG10. Effects of speaking rate on the perceived internal structure of phonetic categories. Joanne L. Miller and Lydia E. Volaitis (Department of Psychology, Northeastern University, Boston, MA 02115)

It is well known that when listeners identify consonants as voiced or voiceless on the basis of voice onset time (VOT), they do so in a rate-dependent manner: As rate of speech decreases such that overall syllable duration increases, the category boundary moves toward a longer VOT value. We have been examining whether this alteration in boundary location is accompanied by a change in the specific stimuli judged to be the best category exemplars. For our experiment we created two /bi/–/pi/ series with overall syllable durations of 125 and 325 ms, respectively. VOT varied from 10–120 ms for the 125-ms series and from 10–320 ms for the 325-ms series. We asked listeners to judge each syllable for its goodness as a member of the /p/ category, using a scale from 1–10; the higher the number, the better the exemplar. For each series, as VOT increased there was an initial increase in rating response, followed by a subsequent decline. Thus for both the short and long syllables, the /p/ category was perceived as having internal structure, with a limited range of stimuli serving as the best exemplars. However, the entire function for the longer syllables was shifted toward longer VOT values. That is, as syllable duration increased, so too did the VOT values of the stimuli judged to be the best category members. Moreover, in a subsequent control experiment we established that the shift in rating funtion was not due to range or frequency effects, but to the alteration in syllable duration, per se. These findings strongly suggest that the adjustment for speaking rate, whatever the underlying mechanism, involves a comprehensive modification in the mapping between acoustical signal and phonetic structure.

3:30

GGG11. Formantless vowels and theories of vowel perception. Frank Gooding (Department of Linguistics, University College of North Wales, Bangor LL57 2UW, United Kingdom)

Most theories of vowel perception rely on a stage of formant peak extraction. However, combining the indications from earlier studies with single formant vowels [Chistovich and Lublinskaya, Hear. Res. 1, 185–193 (1979)] and formantless vowels with "flat" spectra [Carpenter and Morton, Lang. Speech 5, 203–214 (1962)], it would appear that perceived vowel quality does not depend on extraction of two or more formant peaks. In this study, Carpenter and Morton's results are extended by constructing a series of tones with "step" spectra with $F0$ (125 Hz) represented by the first harmonic. It was discovered that a full range of clear, satisfactory front vowel qualities ([i] to [a]) resulted with a band of high-frequency harmonics, where the upper frequency cutoff was 3.5 kHz, and the low-frequency edge varied from 3–1 kHz. Similarly, a back vowel series could be created with a low-frequency band consisting of the first two to the first nine harmonics. The results are interpreted in terms of gross spectral dominance as a prime determinant of phonetic vowel quality. Further experiments with formant and formantless vowels are planned to test this hypothesis.

3:42

GGG12. Using P centers to assess the coherence of syllables. Charles Hoequist, Jr. and Mary R. Smith (Speech Laboratory, Engineering Department, Cambridge University, Trumpington Street, Cambridge CB2 1PZ, United Kingdom)

Across studies when listeners perceive a syllable to have a fricative–stop initial cluster, they report only one P center. Further, the locations of such P centers shift in a one-to-one millisecond manner with changes in pre-vocalic acoustic durations. We also know there is a lower limit on the duration of silence which allows such clusters to be perceived. In this study we ask about the upper limit on intra-syllabic silence and we ask if the number and location of P centers can be used to assess listeners' sensitivity to the segmental coherence of syllables. Using natural tokens of [s], [trip] and [strip] by two talkers, the silent interval was varied from (a) isochronous alternation to (b) overlap of [s] and [trip]. Listeners judged the number of events and the location of P centers. For (a) they perceive two separate events, each with a P center. For (b) they perceive two simultaneous events with one P center. When listeners report one event with one P center, they show sensitivity to the naturally produced silent intervals. [Work supported by Alvey.]

3:54

GGG13. Hemispheric specialization for familiar voice recognition: Dichotic listening evidence. Jody Kreiman (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024) and Diana Van Lancker (Audiology and Speech Pathology, Sepulveda VAMC, and Department of Neuropsychology, Center for Health Sciences, UCLA, Los Angeles, CA 90024)

Previous dichotic listening studies of voice recognition have produced ambiguous results, perhaps due to the use of unfamiliar voices as stimuli: Evidence from brain-damaged subjects [e.g., D. Van Lancker and J. Kreiman, UCLA Working Papers in Phonetics 62, 50–60 (1985)] suggested that both hemispheres are required to discriminate among unfamiliar voices. In contrast, recognition of familiar voices was observed to require an intact right hemisphere in the unilaterally brain-damaged population. To obtain data on ear advantages for dichotically presented voice stimuli, therefore, we used voices of politicians and entertainers familiar to many listeners. Subjects monitored one ear at a time and identified both the speaker and the word spoken on each trial. Although not significant, the ear advantage observed for voice recognition did differ significantly from the significant right ear advantage found for word identification. In light of current models of hemispheric specialization, we interpret this *relative* left ear advantage for voice recognition as a reflection of a right hemisphere superiority for familiar speaker recognition in normal subjects.

4:06

GGG14. Timbre: A better musical analogy to speech? Arthur G. Samuel (Department of Psychology, Box 11A Yale Station, Yale University, New Haven, CT 06520)

A recurring claim in speech research has been that "speech is special"—different perceptual processing is involved for speech than for other acoustic stimuli. In response to this claim, several demonstrations of speech-like results have been reported for nonspeech stimuli, including musical intervals (e.g., major versus minor chords), and manner of playing a string (e.g., pluck versus bow). The research to be reported explores another, possibly more appropriate, musical analog: timbre. The notion is that families of instruments (e.g., horns, strings) may be analogous to families of phonemes (e.g., liquids, stops, fricatives). To test this notion, stimuli were synthesized to sound like various instruments. Standard speech experiments were then conducted, including forced-choice identification, ABX discrimination, duplex perception, selective adaptation, and paired contrast. The similarity of the obtained results to those for speech will be reported, and the implications of these findings for theories of speech perception will be discussed. [Work supported by AFOSR.]

4:18

GGG15. Categorical perception of speech intonation contour: The psychoacoustic basis of tonetic categories. Andrew Faulkner (Speech Research Group, IBM UK Scientific Centre, Saint Clement Street, Winchester, Hampshire SO23 9DR, United Kingdom)

The notion of perceptual categories for intonation contour has particular attractions in speech recognition and speech synthesis. According to the logic of categorical perception, tonetic category boundaries should correspond to discrimination peaks. Can such peaks be found, and if so, is their basis speech-specific or psychoacoustic? A series of ABX experiments will be reported that examine discriminability along stimulus continua formed by one- and two-section logarithmic f_0 slopes, imposed on both synthetic speech, and complex tone stimuli. A single discrimination peak was found, for both speech and tone stimuli, in the region of a flat f_0 slope. This discrimination peak corresponds to a putative category boundary which could distinguish *rise* from *fall* contours, on a one-section slope, and distinguish *fall* from *fall–rise* contours, on a two-section slope. Because the same discrimination peak is evident for speech and non-speech stimuli, the basis of the category boundary effect can be regarded as psychoacoustic.

Session HHH. Underwater Acoustics XI: Scattering

Darrell R. Jackson, Chairman
Applied Physics Laboratory, University of Washington, Seattle, Washington 98195

Contributed Papers

2:00

HHH1. Some concepts in the statistical energy analysis. G. Maidanik (David Taylor Naval Ship Research and Development Center, Bethesda, MD 20084-5000)

In a recent presentation an attempt was made to introduce basic concepts in the development of the elements of the statistical energy analysis (SEA). [G. Maidanik, J. Acoust. Soc. Am. Suppl. 1 **79**, 511 (1986)]. The analytical definitions and manipulations were exercised on a simple dynamic system so that they could not clutter unduly the conceptual understanding of SEA. Moreover, although SEA is devised to deal mostly with the dynamic behavior of interacting structural system, many of the basic concepts that underlie SEA can be explained and understood employing a single noninteracting dynamic system. The modal approach was used and the division of the energy stored in the dynamic system into a resonant and a nonresonant part was derived. The significance of this division was briefly discussed. This paper discusses more recent concepts such as the division of stored energy into a direct and a reverberant part.

2:15

HHH2. Sound scattering by fluid cylinders of finite length. T. K. Stanton (Department of Geophysics, University of Wisconsin, Madison, WI 53706)

The scattering of incident plane waves by penetrable fluid finite-length circular cylinders for all frequencies is described. The results appear to be valid for all frequencies for lengths much greater than the diameter of the cylinder and limited to low frequencies ($ka \ll 1$) when the length is comparable to the diameter. By neglecting end effects, we are able to approximate the volume flow per unit length of the scattered field of the finite cylinder by that of the infinite cylinder. The solution is obtained by integrating this volume flow along the length of the cylinder. This approximation restricts the solution to geometries where the direction of the incident wave is normal or near-normal to the lengthwise axis of the cylinder. The results are compared to those from other theories and to data. Under the same limiting conditions that the other theories were derived, the solution in this paper is identical to those. There was good to excellent agreement between this solution, when appropriately truncated to include just the first few modal terms, and lead cylinders and preserved shrimp. Finally a simple heuristic "high-pass" model is also derived and shown to pass through the data and asymptotically approach the rigorous solution at extreme values of ka. [Work supported by ONR.]

2:30

HHH3. Necessary conditions for superresonant interaction between monopole scatterers. I. Tolstoy (Knockvennie, Castle Douglas, SW Scotland)

Systems of precisely spaced bubbles, balloons, or air-filled thin shells in water, in full spaces, or near elastic boundaries, insonified at frequencies close to the fundamental resonance value ω_0 ("bubble frequency") and interacting *via* multiple scatter may develop true resonant modes or superresonances (SRs). Under SR conditions the pressure amplification relative to the incident field will be of order $(k_R a)^{-2}$, as opposed to $(k_R a)^{-1}$ for single scatter at ω_0, k_R being the wavenumber in water and a the scatterer radius. For simple configurations this leads to theoretical SR amplifications between 10^3 and 5×10^3. It is shown here that, in order to observe the SR effect, spacings and volumes must be controlled to $\frac{1}{2}\%$ or better. Typical pair spacings are of order $\frac{1}{2}\lambda_{\rm acoust}$ or (roughly) multiples thereof. When a bubble/balloon pair is near a thin plate the basic spacing is about one flexure-wave wavelength and SR amplification becomes very sensitive to the direction of flexure mode arrivals, tending to vanish entirely for angles intermediate between broadside and endfire incidence. [Work supported by ONR.]

2:45

HHH4. Transient and steady-state target resonance vibrations extracted from their acoustic echoes. G. C. Gaunaurd and C. Y. Tsui (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20903-5000)

Scattering experiments carried out decades ago [W. Angeloff and F. Abbot, NEL TR-1273 (Mar. 1965)] are examined in the light of modern theoretical advances. This places the early observations on solid theoretical foundations and permits a reliable and understandable measurements program. This description is achieved by the suitable sampling and processing of the returned signals. It was found that pulses backscattered from targets are divided into three parts, an initial transient region, a steady-state region, and a final transient regime corresponding to the reradiation of the stored energy by the structure. If the backscattered pulse is sampled in the steady-state or free-vibration region, one obtains the usual backscattering cross section of the body. However, if the sampling is done within the final transient regime, near the tail end of the pulse, then the "backgrounds" of the resonance scattering theory (RST) are automatically suppressed and one obtains, *experimentally*, the "spectrogram" of the body, with its modal resonances completely isolated. In this case, if one sweeps the angle θ around the body with the frequency fixed at the value of any resonance frequency, one then obtains "rosetta" patterns, which yield graphic multipole interpretations of the resonant modes excited. Theory and experiments match well, and we exhibit many examples.

3:00

HHH5. Nonlinear distortion of the bubble pulse in a layered inhomogeneous ocean. David Epstein (Science Department, SUNY Maritime College, Bronx, NY 10465)

Many years ago, the author analyzed a representative deep-fired long-range underwater explosion waveform [J. Acoust. Soc. Am. **35**, 800 (1963)]. The most striking feature was the so-called "double shock" formation, e.g., a strong resemblance between the shock wave and bubble pulse in both amplitude and shape. The phenomenon is attributed to the combined effects of detonation depth and waveform propagation. In a previous paper the characteristics of an explosion waveform at the source, as function of shot depth was established [Proc. 11th ICA, Paris (1983), Vol. 1, pp. 345–348]. Weak shock theory was used to study nonlinear distortion of the bubble pulse as it propagates in an homogeneous, unbounded, dissipationless medium [Proc. 10th Int. Symp. Nonlinear Acoust., Kobe, Japan (1984), pp. 79–82]. It was found that the deepest fired shots suffer the greatest distortion but that, in general, extremely long range is required for shock wave formation. However, focusing due to inhomogeneous structure may substantially increase pulse strength thereby enhancing nonlinear effects. Here the effect of inhomogeneous layering on the range required for shock wave formation is investigated.

3:15

HHH6. Resonance identification through bistatic scattering for elastic spheres and spheroids. H. Überall (Department of Physics, Catholic University, Washington, DC 20064), M. F. Werby (NORDA, Code 221, NSTL, MS 39529), S. H. Brown, and J. W. Dickey (David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402)

Free vibrations of a submerged elastic object, resonantly excited by an incident acoustic wave or wave train, occur in the characteristic fashion of one of an infinite series of allowed eigenvibrations (normal modes for objects of separable geometry) if the excitation takes place at the eigenfrequency of that resonance. While such resonances appear in the backscattering cross section plotted versus frequency, interfering with the contribution of specular echoes, only resonance frequencies and resonance widths can be determined from such a plot but not the resonance order. For targets of separable geometry it has been shown both experimentally [G. Maze and J. Ripoche, J. Acoust. Soc. Am. **73**, 41 (1983)] and theoretically [M. F. Werby and H. Uberall, submitted to J. Acoust. Soc. Am.] that bistatic observations can determine the resonance order after subtraction of the specular background, and the dipole resonance of the Rayleigh wave has been theoretically identified in this way. The method is here applied to bistatic scattering from elastic spheroids, as analyzed by the *T*-matrix formalism and computer code. [H. Überall is supported by the David W. Taylor Naval Ship Research and Development Center, and by the Office of Naval Research.]

3:30

HHH7. Amplitude and spectral characteristics of low grazing angle backscattering. Paul D. Koenigs (Naval Underwater Systems Center, New London, CT 06320-5994)

Acoustic backscattering from the sea surface at grazing angles in the region of 5 deg is affected by shadowing. Under these conditions the equations presented by Bass *et al.* [IEEE Trans. Antennas Propag. **AP-16** (5) (Sept. 1968)] may be combined to form a single equation. The agreement between experimental data near 1 kHz and the unified equation is good. The frequency shift of the backscattered acoustic energy is dependent on wave height and period through the horizontal orbital particle velocity. The frequency spread of backscattered energy is still dependent on the same particle velocity but is diminished because the wave troughs are preferentially insonified. The experimentally determined values of backscattering strength compare favorably with other reported low-frequency, low grazing angle data.