

Comparison of procedures for determination of acoustic nonlinearity of some inhomogeneous materials

Jensen, Leif Bjørnø

Published in:
Acoustical Society of America. Journal

Link to article, DOI:
[10.1121/1.2020879](https://doi.org/10.1121/1.2020879)

Publication date:
1983

Document Version
Publisher's PDF, also known as Version of record

[Link back to DTU Orbit](#)

Citation (APA):
Jensen, L. B. (1983). Comparison of procedures for determination of acoustic nonlinearity of some inhomogeneous materials. *Acoustical Society of America. Journal*, 74(S1), S27-S27. DOI: 10.1121/1.2020879

DTU Library

Technical Information Center of Denmark

General rights

Copyright and moral rights for the publications made accessible in the public portal are retained by the authors and/or other copyright owners and it is a condition of accessing publications that users recognise and abide by the legal requirements associated with these rights.

- Users may download and print one copy of any publication from the public portal for the purpose of private study or research.
- You may not further distribute the material or use it for any profit-making activity or commercial gain
- You may freely distribute the URL identifying the publication in the public portal

If you believe that this document breaches copyright please contact us providing details, and we will remove access to the work immediately and investigate your claim.

PROGRAM OF

The 106th Meeting of the Acoustical Society of America

Town and Country Hotel • San Diego, California • 7-11 November 1983

TUESDAY MORNING, 8 NOVEMBER 1983

SENATE/COMMITTEE ROOMS, 8:30 A.M. TO 12:10 P.M.

Session A. Underwater Acoustics: Arctic Acoustics I

William Mosely, Chairman
Naval Research Laboratory, Washington, DC 20375

Chairman's Introduction—8:30

Invited Papers

8:35

A1. Bottom-interacting acoustic signals in the Arctic channel: Long-range propagation. Henry W. Kutschale and Tai Lee (Lamont-Doherty Geological Observatory of Columbia University, Palisades, NY 10964)

Hydroacoustic signals from underwater explosions that have propagated over the Arctic abyssal plains commonly display marked frequency dispersion in pulses that are bottom-interacting and that arrive after the SOFAR signal. In the infrasonic band of 2 to 20 Hz, the temporal dispersion for each pulse that has interacted with the flat bottom of the plain can be nearly as strong as that observed in the SOFAR signal for the first mode. However, the bottom-interacting pulses correspond to a coherent summation of many higher-order normal modes in a channel bounded above by the ocean surface and below by the upper 400 m of the bottom sediments, where the velocity increases with depth. Using normal-mode theory and the Multiple Scattering Pulse Fast Field Program (MSPFFP), we have analyzed the dispersion and pulse shapes and have derived the acoustic properties of the bottom in the Pole, Barents, and Mendeleev Abyssal Plains. The principal properties of the bottom controlling the propagation are compressional velocity, density, and attenuation. In contrast, the ice layer has a negligible effect on the dispersion of the observed waves. The effect on pulse compression of this frequency dispersion of the bottom-interacting signals was simulated numerically, using predistorted waveforms matched to the dispersion of the SOFAR channel at specified ranges.

9:00

A2. Ambient noise in the central Arctic Ocean. Ira Dyer (Department of Ocean Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Underwater ambient noise has been measured in the central Arctic Ocean on three separate occasions with the use of a horizontal and a vertical array. These data help to identify various noise mechanisms. Of principal interest is noise caused by various ice cracking events which depend upon stress levels in the ice and which seem to have varying temporal and spectral characteristics. I review these mechanisms and show how the data relate to them. While spatial analyses have been carried out only at very low frequencies, spectral analyses cover the range from 1 to 10 000 Hz. In this frequency range noise mechanisms other than pack ice are sometimes important, such as wind-generated noise and earthquake noise. [Work supported by ONR.]

9:25

A3. Seismic/acoustic propagation in the Arctic Ocean. Arthur B. Baggeroer and Gregory L. Duckworth (Departments of Ocean and Electrical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

During the FRAM II and FRAM IV experiments in the Pole Abyssal Plain and the Barents Abyssal Plain of the Arctic Ocean 24 channel hydrophone arrays approximately 1 km × 1 km in extent were used to record digitally both seismic refraction and long-range acoustic propagation data. From these data the velocity versus depth function for the seismic structure along the FRAM II and FRAM IV drift tracks were estimated by high-resolution velocity spectral analysis and tau-slowness migration methods. These indicate oceanic layers 2 and 3, but they are relatively thin. These techniques were also used to identify internal and surface-reflected

multipaths as well as converted shear paths to obtain a detailed characterization of the seismic propagation. The long-range acoustic signals contained strong modal components and bottom interacting ones. Velocity spectra and autoregressive algorithms resolved the dispersive properties of the phase and group velocity of each mode. These were used to estimate the water column sound velocity profile with an inversion algorithm. During FRAM II the bottom interactions lasted up to 80 s long and contained as much energy as the low-order modes at low frequencies. These signals were used to estimate the shallow velocity versus depth structure. During FRAM IV the dispersion characteristics of the low-order modes changed slightly. In contrast, virtually no bottom-interacting energy was observed suggesting strong scattering of the deep paths by an intervening midoceanic ridge. [Work supported by ONR Arctic Program Office.]

9:50

A4. TRISTEN/FRAM IV cw spatial coherence and temporal stability. F. R. DiNapoli, R. Nielsen, D. Potter (Naval Underwater Systems Center, New London, CT 06320), and P. L. Stocklin (Analysis and Technology, Inc., N. Stonington, CT 06359)

The TRISTEN-82/FRAM IV experiment was a multifaceted experiment investigating acoustics in the Arctic. This paper focuses on the spatial coherence and temporal (frequency) stability of acoustic signals transmitted and received between fixed ice camps separated by approximately 130 nmi. A high-powered, low-frequency (NUSC HLF-3 Arctic) hydroacoustic source transmitted stable cw tones of 1 h or more at various frequencies from 5 to 200 Hz during April 1982. These signals were received on an X-shaped array having an aperture of 1200 m on each leg. The array was operated by the Massachusetts Institute of Technology and the Wood Hole Oceanographic Institute. The spatial coherence, both broadside and end-fire to the source were found from normalized sensor pair cross correlation, corrected for signal-to-noise ratio. Total and 3-dB down received signal bandwidths were found using complex demodulation and FFT with a resolution to a fraction of mHz, followed by cumulative energy analysis in the frequency domain. The experimental setup data, analysis procedures, and results will be presented. [Work sponsored by the Office of Naval Research, Code 425-AR.]

10:15

A5. Arctic data collection the easy way—Data buoys. Beaumont M. Buck (Polar Research Laboratory, Inc., 123 Santa Barbara Street, Santa Barbara, CA 93101)

This paper presents a review of developmental and operational use of Arctic data buoys. There are many problems involved with aircraft and icebreaker logistics for scientific support in the Arctic. Manned research stations on sea ice are expensive, self-contaminating in terms of acoustic and electromagnetic noise, and generally limited to a 40- to 50-day period in the springtime. Autonomous data buoys, on the other hand, do not suffer those limitations. These devices can be installed by paradrop, aircraft ice-landing, submarine, or icebreaker in almost any geographic area of the Arctic Ocean, peripheral seas, and marginal sea ice zones to collect and report data over all seasons. There are restrictions, of course, as to what can be accomplished automatically within a small buoy hull, and almost all Arctic buoys, so far, have been mechanically passive. Recent advances in low-power microprocessors and memories enable considerable increases in the extensive in-buoy processing needed for data compression to accommodate the limited capacities of polar orbiting scientific satellites. [Work supported by U.S. Navy and NOAA.]

Contributed Papers

10:40

A6. Normal mode and ray equivalence in the Arctic sound channel. Robert H. Mellen (B-K Dynamics, Inc., 247 Shaw Street, New London, CT 06320)

Effects of upward refraction combined with scattering by the rough ice interface result in a dispersive sound channel capable of propagating only very low frequencies to long range. Because only a few modes are involved, normal mode theory is appropriate. However, surface-scatter models are usually based on plane-wave approximations for which ray theory is most useful. The two approaches are simply combined in the impulse-response method. Application of this method to the propagation of transients and continuous wave signals is discussed. Removal of dispersive effects by deconvolution to increase data rate is also considered in view of the extreme temporal stability of the channel. [Work supported by ONR and NUSC.]

10:55

A7. Reverberation model of the arctic channel in the presence of ridge scattering. Garry M. Jacyna and Alan J. Friedman (Planning Systems Inc., McLean, VA 22102)

A ray-theoretic model is advanced to describe the average reverberation intensities within an Arctic channel bounded by a pack-ice surface and an upward-refracting sound-speed profile in the presence of pressure ridge scattering. The surface is assumed random and a phenomenological model based on Kirchoff assumptions in the high-frequency limit is derived. Relevant parameters are estimated from data supplied by Chapman and Scott using a curve-fit routine. Ridges are modeled as inverted cones, the surfaces of which are also assumed random. This simplifying geometric assumption allows for a highly tractable technique. Forward as well as backscattered geometries are treated with no assumptions regarding source/receiver range or depth and ridge location or draft. Applications to sonar receiver performance will be discussed in a companion paper. [Work supported by ONR.]

11:10

A8. Sonar receiver performance models for the arctic channel in the presence of ridge scattering. Garry M. Jacyna and Alan J. Friedman (Planning Systems Inc., McLean, VA 22102)

A generic class of sonar receivers are modeled in an attempt to describe average receiver performance in the Arctic channel. A ray-theoretic model, discussed in a companion paper, describes the average reverber-

ation intensities within an Arctic channel in the presence of pressure ridge scattering. Receiver performance is evaluated in terms of signal detectability as a function of range, ridge location, and source/receiver depth. The models are extremely general in that arbitrary pulse shapes and sensor locations are allowed. As an illustration, two pulse shapes are considered—a transient and a CTFM pulse. Their effect on receiver performance is dependent on the source/receiver geometry as well as ridge location. In general, however, the longer CTFM pulse often leads to a reverberation-limited condition as a result of surface as well as pressure ridge scattering. [Work supported by ONR.]

11:25

A9. Depth dependence of the temporal characteristics of cw propagation in the Arctic Ocean. G. R. Giellis, T. C. Yang, and C. W. Votaw (Code 5123, Naval Research Laboratory, Washington, DC 20375)

Previous work by P. Mikhalevsky [J. Acoust. Soc. Am. 70, 1717–1722 (1981)] investigated the temporal characteristics of cw signals propagating in the Arctic Ocean. The previous work was limited to receivers at a depth of 91 m. In this paper, the temporal characteristics will be extended to 28 hydrophones in a vertical array with depths from 30 to 960 m. The data were taken during the FRAM IV experiment in April 1982. The signals were generated at the Tristen ice camp and received at the FRAM IV ice camp located north of Spitzbergen. The range of approximately 275 km was over the Mid-Ocean Ridge; the average water depth along the path was 3000 m. Most of the data examined are at a high signal-to-noise ratio (>10 dB) which makes it possible to study the temporal fluctuation and the statistics of the cw signals at different depths. By employing a mode filtering technique, the temporal dependence of the amplitude and phase of the lowest order mode has been investigated. The drift rates of the ice

camp was negligible during the transmission periods and hence the results apply to transmission between essentially fixed stations.

11:40

A10. Ultrasonic reflection measurements of floating ice. Neal G. Brower (Applied Physics Laboratory, Johns Hopkins University, Laurel, MD 20702), Kin W. Ng, and Walter G. Mayer (Department of Physics, Georgetown University, Washington, DC 20057)

An experimental investigation of reflection of ultrasonic, bounded beams from floating ice has been conducted. Ultrasonic frequencies and ice thicknesses corresponding to frequency thicknesses of between 2 and 5 mm-MHz are considered. Nonspecular reflection profiles are obtained using schlieren techniques. These nonspecular profiles are compared to theoretical profiles calculated for water/ice plate/air systems. [Work supported by ONR.]

11:55

A11. Acoustic scattering from Arctic sea-ice ridges using high-angle PE. Robert R. Greene (Science Applications, Inc., 8400 Westpark Drive, McLean VA 22101)

Long-range propagation loss runs with the high-angle PE model indicate that scattering of sound into the ocean bottom by sea-ice roughness can account for the frequency dependence of transmission loss in the Arctic. But differences between the slopes of the data and predicted transmission loss curves suggest the presence of an additional loss mechanism in the ice.

TUESDAY MORNING, 8 NOVEMBER 1983

CALIFORNIA ROOM, 9:00 A.M. TO 12:00 NOON

Session B. Noise I: Use and Misuse of the FFT

John C. Burgess, Chairman

Department of Mechanical Engineering, University of Hawaii, 2540 Dole Street, Honolulu, Hawaii 96822

Chairman's Introduction—9:00

Invited Papers

9:05

B1. Digital signal processing for sound and vibration analysis: A tutorial. John C. Burgess (Department of Mechanical Engineering, University of Hawaii, 2540 Dole Street, Honolulu, HI 96822)

The literature on digital signal processing is based on four kinds of transform: Fourier transform (FT), z-transform, Fourier series, and discrete Fourier transform (DFT). Digital implementation uses only the DFT, usually in the form of an FFT algorithm. Some popular texts attempt to describe digital results by using analog (usually FT) analysis. Some have presented analytical results for infinite length random signals and then used illustrations based on finite length periodic signals. What "windows" do and their different effects on random and periodic signals seldom are clearly identified. There are two implementations of the DFT that differ by a constant factor. Fast convolution and correlation analyses require special handling to avoid "circular" effects. The DFT treats each finite length sample from a random signal as if it were a sample from a deterministic signal. The purpose of this tutorial is to illuminate these and other potentially troublesome aspects of digital signal processing pertinent to applications in sound and vibration analysis.

9:35

B2. Noise source identification by coherence and acoustical methods using FFT analyzers. Thomas H. Hodgson (Center for Sound and Vibration, North Carolina State University, Raleigh, NC 27650)

The practical understanding of the two-channel FFT spectrum analyzer as used in noise source identification will be addressed. In particular, the measurement of the coherence function and the acoustic intensity, the

latter by both two-microphone and surface methods, will be described. Problems related to amplitude and phase information and the design of most analyzers as they relate to Fourier transform theory or electrical network practice will be discussed. Several case studies of noise source identification measurement will be presented in order to demonstrate some of the points related to FFT analyzer use described in the paper. Both steady-state and impact noise problems will be considered with reference to causes of possible errors.

9:55

B3. Field experience with sound and vibration measurements using an FFT analyzer. Frank H. Brittain (Bechtel Research and Engineering, P.O. Box 3965, San Francisco, CA 94119)

The theory upon which FFT analyzers are based promises a very powerful and useful measurement tool. In the field, a variety of practical difficulties often affect the ability to obtain valid results using an FFT analyzer. These difficulties fall into five categories: (1) inadequate knowledge on the part of the user, (2) details of the analyzer as implemented by the manufacturers, (3) bugs in the hardware and software of the analyzer, (4) conditions under which measurements are made (including the individual characteristics of the equipment tested and the background vibration), and (5) ability of the user to interpret data from the analyzer. Each of these five categories is discussed, with emphasis on two-channel measurement applications. Practical examples of field problems encountered and methods used to overcome them are discussed.

10:15

B4. Status of standards on digital systems for sound and vibration analysis. Tony F. W. Embleton (Division of Physics, National Research Council, Ottawa, Ontario K1A 0S1, Canada)

Existing ANSI (American National Standards Institute) standards deal primarily with analog acoustical instruments and their use—following traditional thinking and procedures of acousticians. They are relevant to the analog parts of a digital signal processing system such as the signal itself (time and frequency characteristics), transduction (directionality, linearity, frequency response of microphones or vibration pick-ups), and output (preferred frequencies, reference quantities). They are also relevant to digital components that mimic closely the behavior of analog devices, e.g., filters that receive signals continuously. The next revision of S1.11 will explicitly include specifications for the performance of both analog and digital filters. The IEC (International Electrotechnical Commission) acoustical standards similarly relate to analog instruments and their use. The next several years should see progress in the development of standards for digital acoustical instrumentation, but before this can happen acousticians must recognize and widely discuss the technical problems that exist. General consensus should be reached on many factors, one of the most fundamental is the possible methods of achieving a desired degree of accuracy for a given kind of acoustical signal. Standards written prematurely risk biasing measurement procedures, and instrumentation to meet these needs, in what may be less-than-optimum directions.

10:35 to 11:00

Panel Discussion

Contributed Papers

11:00

B5. Amplitude calibration of an FFT analyzer. Joseph Pope (Bruel & Kjaer Instruments, Inc., 185 Forest Street, Marlborough, MA 01752)

When used for frequency analysis, an FFT analyzer produces a line spectrum on the amplitude-frequency plane. Each spectrum line represents the output of a filter centered at the frequency associated with the line, the shape of the filter being determined by the effective time window applied to the data being analyzed. While it is generally recognized that the use of an explicit window function is often desirable—e.g., to control spectral “leakage”—many FFT users seem unaware that *use of a window implies a compensatory calibration adjustment that is dependent upon the character of the data and the objective of the analysis*. Effective filter shapes for rectangular, Hanning, flat top, and Kaiser-Bessel windows are discussed along with the concept of filter amplitude response and effective noise bandwidth. Appropriate calibration for rms, power, power spectral density, and energy spectral density spectra is explained.

11:15

B6. Acoustic impedance measurement via FFT. Werner G. Richarz (Institute for Aerospace Studies, University of Toronto, 4925 Dufferin Street, Downsview, Ontario, Canada)

The measurement of acoustic impedance of acoustic materials and networks often requires a rather laborious impedance tube measurement.

Other methods have been evolved, but these usually rely on two microphone measurements. The *single microphone* system reported here utilizes the incident and reflected pulse in a circular impedance tube. The pulse is generated by a suitably damped loudspeaker driven by a low-frequency square wave. A two-channel Fast Fourier Transform analyzer captures the pressure-time history. The signal is then conditioned leaving only replicates of the incident and the reflected pulses suitably shifted in time to simulate conditions at the sample surface. Using the various features of modern FFT analyzers, the reflection (absorption) coefficient as well as the complex valued impedance are readily available from a *single* measurement. At present the usable frequency range is 5 kHz. [Work supported by the Natural Science and Engineering Council, Canada.]

11:30

B7. Use and misuse of an FFT analyzer for octave-band and one-third octave-band analysis. Richard C. Sohaney (Bruel & Kjaer Instruments, Inc., 185 Forest Street, Marlborough, MA 01752)

An FFT analyzer is often used as a “real time analyzer” to perform octave-band and one-third octave-band analyses. The limitations and assumptions which the filter synthesizing technique imposes on the data signal are discussed. A method for synthesizing ANSI Class II octave-band and Class III one-third octave-band filters using an FFT analyzer is explained with emphasis on the shortcomings of synthesized filters. Example analyses of various signals are performed with conventional ANSI

Class III one-third octave-band filters and an FFT analyzer to demonstrate correct and incorrect use of the synthesized filter method.

11:45

B8. Data averaging and FFT devices. David K. Holger (Department of Engineering Science and Mechanics, 214 ERI Bldg., Iowa State University, Ames, IA 50011)

FFT devices often include a data smoothing capability that is based upon averaging over a selected number of adjacent points in a data array. Averaging time domain data in this way is equivalent to low-pass filtering the data. The resulting inaccuracies associated with frequency domain representations of the averaged data are examined. Predicted and experimental results are presented for the FFT's of averaged sinusoidal, periodic, and filtered random waveforms. The effects of sample interval and number of averaging points on the equivalent filter characteristics of the averaging process are discussed.

TUESDAY MORNING, 8 NOVEMBER 1983

COUNCIL/CHAMBER ROOMS, 9:00 TO 10:40 A.M.

Session C. Engineering Acoustics I and Architectural Acoustics I: Teleconferencing

James E. West, Chairman
Bell Laboratories, Murray Hill, New Jersey 07974

Chairman's Introduction—9:00

Invited Papers

9:05

C1. Acoustics of teleconferencing. Chris Stockbridge (American Bell Inc., E.D.&D., Room 4G-514, Holmdel, NJ 07733)

Room acoustics is an element of teleconferencing too often neglected. Ideally, the room should not be symmetrical and the walls should be nonparallel, covered irregularly with sound scattering and noise absorbing panels. At the very least, a place for teleconferencing must be quiet and not reverberant. Technical specification of various sizes of desirable acoustical spaces which match both people's behavior and equipment constraints will be set forth in this talk. Equipment and techniques will be discussed which select acoustics for a natural feel within the room while generating signals that create clear and recognizable voice sounds when heard at remote location(s). These include phased array directional microphones, carefully positioned loudspeakers, simple acoustic absorption, and for the more sophisticated, nonparallel walls and irregular ceiling clouds. The real challenge is to implement this technology in an aesthetically pleasing fashion, tailored and matched to the culture of the expected user population. A variety of teleconferencing rooms will be presented in photographs, showing evolution in the design of rooms for audio, audiographic, and full motion video teleconferencing.

9:30

C2. The human engineering of a new teleconferencing service. Daryl J. Eigen (Bell Laboratories, Naperville-Wheaton Road, Naperville, IL 60566)

The Bell System is developing a new network teleconferencing service. This service enables the customer to set up a conference with or without an operator, and it incorporates state-of-the-art signal processing technology to enhance audio quality. The human engineering of this new service concentrated on characterizing the customer, optimizing the perceived audio quality, and making the conference setup procedures easy to use and error free. A coordinated series of interviews, surveys, laboratory studies, and field studies was implemented to increase customer satisfaction, performance, and usage. This sequence of testing is culminated in a Controlled Product Test, which is a dress rehearsal of a new service, using Bell System employees as the first customers. The CPT started on 7 February 1983 in selected sites in San Diego, Los Angeles, and San Francisco with two bridges located in Los Angeles. The CPT has since been expanded to additional sites in 12 states with four more bridges, two in White Plains, New York and two in Chicago, Illinois. This geographic diversity provides the opportunity for large calling volumes and a wide range of conference configurations. Over 2000 calls have been placed thus far, allowing the conference setup procedures to be simplified and perceived audio quality to be improved.

Contributed Papers

9:55

C3. The Quorum TM teleconferencing microphone. James H. Snyder (American Bell, Holmdel, NJ 07733)

Arrays of microphones have long been used in underwater acoustics,

but only recently has it been appreciated that the directivity of microphone arrays makes them superior to conventional microphones in many audio applications as well. In this paper we describe a particular microphone array, the Quorum TM teleconferencing microphone, whose directivity is achieved by the nonuniform spacing of the transducers. We pres-

ent equations for the array sensitivity assuming plane and spherical wave radiation, and compare these results with measurements.

10:10

C4. Conference microphone with adjustable directivity. J. L. Flanagan and R. A. Kubli (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

Designs are given for fixed-steered line-array microphones suitable for teleconference applications. The steering is arranged so that the array can be located out of line-of-sight of the conferees. The directivity pattern of the transducer can be selected to give effective spatial coverage for a variety of conference conditions. Additionally, the designs utilize acoustic summation and anti-aliasing filtering. They require only an inexpensive electret element as the transducer. Theoretical responses are calculated and plotted on an Apple II computer. A prototype model is fabricated, and experimental measurements, made under free-field conditions, are shown to compare satisfactorily with the theoretical responses.

10:25

C5. An acoustic direction finder. C. H. Coker (Bell Laboratories, Murray Hill, NJ 07974) and D. R. Fischell (Bell Laboratories, Holmdel, NJ 07733)

A simple procedure is described for finding the direction to the source of an acoustic signal—primarily the human voice. The method, in effect, constructs the cross correlation of severely distorted signals from two microphones. The distortion, a combination of time-domain processing and digital logic, is such to give strong preference to direct-path early arrivals over reflections that follow, and to produce streams of pulses whose cross correlations can be computed very simply. The method has been implemented as a combination of custom hardware, and a simple microprocessor. The implementation operates in real time (with ~0.2-s delay) and produces outputs to indicate either the azimuth angle, or inter-microphone delay. Possible uses include self-aiming directional microphones, automatic aiming or switching of video cameras, and remote identification of talkers in nonvideo teleconferencing.

**Session D. Architectural Acoustics II: The Technical Committee on Architectural Acoustics
V. O. Knudsen Distinguished Lecture**

Session D moved to Thursday Afternoon, 10 November 1983 at 1:30 P.M.

TUESDAY MORNING, 8 NOVEMBER 1983

SUNRISE ROOM, 9:00 A.M. TO 12:05 P.M.

Session E. Physiological Acoustics I: Peripheral Systems

Joseph E. Hind, Chairman

Department of Neurophysiology, Medical Science Building, University of Wisconsin, Madison, Wisconsin 53706

Chairman's Introduction—9:00

Contributed Papers

9:05

E1. Comparative cochlear morphology in echolocating cetaceans. D. R. Ketten, F. L. Starr,¹ and D. Wartzok (Department of Immunology and Infectious Disease, John Hopkins Medical Institute, 615 N. Wolfe Street, Baltimore, MD 21205)

The anatomy of the cochlea of six species of odontocetes, or toothed whales (*Physeter catodon*, *Grampus griseus*, *Lagenorhynchus albirostris*, *Phocoena phocoena*, *Tursiops truncatus*, and *Stenella longirostris*) was compared using conventional micrography and radiographic techniques including edge enhancement x-ray, CT scan, magnification radiography, and digital subtraction. These species were selected for differences in fre-

quency and patterning of normal and ultrasonic vocalizations in their natural environments. The cochlea were extracted post-mortem and preserved by injection. Whole cochlea were first examined radiographically for species-specific differences in gross morphology and topology, particularly for angular change of the scalae and torsion of membranous and neural components. Specimens were then decalcified and processed for SEM or thin section microscopy. Measurements obtained by both methods are presented and the advantages of the radiographic techniques discussed. The results are compared with the cetacean cochlear models of Wever [Wever *et al.*, Proc. Natl. Acad. Sci. **68**, 2381–2385 (1971)] and Fleischer [G. Fleischer, J. Paleon. **50** (1), 133–152 (1976)] and analyzed according to the dimension-frequency correlation techniques of Greenwood [D. D. Greenwood, J. Acoust. Soc. Am. **33**, 1344–1356 (1961)] and Hinchcliffe and Pye [R. Hinchcliffe and A. Pye, Int. Audiol. **7**, 259–288 (1968)]. [Work supported by NSF.]^a Also at Radiology Research Laboratory, Johns Hopkins Hospital, Baltimore, MD 21205.

9:20

E2. Three-dimensional directional sensitivity of goldfish 8th nerve fibers. Richard Fay, Sheryl Coombs, and Joseph Baumann (Parmlly Hearing Institute, Loyola University of Chicago, 6525 N. Sheridan Road, Chicago, IL 60626)

Three-dimensional maps of directional sensitivity were constructed for single fibers of the saccular, lagenar, and utricular nerves by measuring acceleration thresholds to sinusoidal motion oriented along 40 axes in three orthogonal planes. Stimuli were generated by adding three mutually perpendicular vibration inputs to a rigid cylinder containing water (and the fish). The phases and amplitudes of the inputs were computed to achieve linear motion vectors with particular orientations, as determined by an accelerometer array. Phase-locking was used to define a cell's acceleration sensitivity, and the polarity of the response (the excitatory direction of motion). In general, the 3-D directional sensitivity of cells (in dB) is described by a solid composed of two tangent spheroids, with an axis of greatest sensitivity passing through their centers. In the saccule, these axes are oriented primarily dorsal-ventrally with deviations from vertical of up to 40°. Lagenar fibers show similar patterns but with greater variation of orientation in the mid-sagittal plane. Utricular units show greater variation in all planes. Any given axis of particle motion is represented in the profile of response across arrays of fibers of different orientation, and is best defined by fibers oriented with a null plane parallel to the axis of motion. [Supported by the NSF.]

9:35

E3. Whole-nerve response to tone bursts in the alligator lizard. Robert G. Turner, Neil T. Shepard, and Donald W. Nielsen (Otolological Research Laboratories, Henry Ford Hospital, Detroit, MI 48202)

The peripheral auditory system of the alligator lizard (*Gerrhonotus multicarinatus*) has been studied extensively; however, limited data are available for the whole-nerve (AP) response. Could the AP serve as a convenient monitor of the condition of the ear during experimentation? AP responses to tone burst acoustic stimuli of various frequencies were recorded from the alligator lizard and three measures using the AP response were obtained. (1) Averaged suprathreshold AP responses were related to primary auditory nerve fiber activity. (2) AP thresholds were compared to single-unit thresholds. (3) AP excitatory tuning curves were compared to single-unit tuning curves. The good agreement between AP responses and single-unit data indicate that the AP is a useful tool for measuring the physiology of the peripheral auditory system in the alligator lizard.

9:50

E4. Linear gain and phase in seismic neurons. Edwin R. Lewis (Electronics Research Laboratory, University of California, Berkeley, CA 94720)

In nearly quiescent seismic environments, many bullfrog (*Rana catesbeiana*) saccular afferent axons exhibit ongoing activity, typically ranging from 10 to 80 spikes/s. Low-level vibration of the whole animal [see H. Koyama *et al.*, Brain Res. **250**, 168–172 (1982) for methods] produces

linear modulation of this activity, allowing one to define small-signal linear transfer ratios and to observe tuning curves for such axons. Each tuning curve to date reflects multiple resonances, sometimes coalescing with no intervening antiresonances, sometimes separated by simple antiresonances. The latter typically are much sharper than the resonances on either side, indicating that they are not simply concomitants of additive parallel resonances. Phase lag typically increases by two cycles or more with increasing frequency through the tuning range (roughly 10 to 200 Hz), with abrupt half-cycle declines through the antiresonances. Resonant and antiresonant frequencies vary from axon to axon, usually lying between 20 and 120 Hz. The amplitudes of peak transfer ratios vary approximately from 50 to 200 spikes/s per cm/s². At 100 Hz, where it occasionally is found, the latter corresponds to approximately one spike/s response for whole-animal displacement of 10 pm. [Supported by NSF Grant BNS-8005834.]

10:05

E5. Filter characteristics of low-frequency cochlear nerve fibers as determined by synchrony response patterns to two-component signals. Steven Greenberg, C. Daniel Geisler, and L. Deng (Department of Neurophysiology, University of Wisconsin, 1300 University Avenue, Madison, WI 53706)

The tuning and filter characteristics of cochlear nerve fibers, as conventionally determined by stimulation with single-component signals, are usually intensity level-dependent. Shifts in characteristic frequency and significant deterioration of frequency selectivity are often observed at supra-rate-saturation intensities. In the present study relatively level-invariant estimates of characteristic frequency (CF) and frequency selectivity were obtained for low-frequency (< 3 kHz) cochlear fibers in the cat via stimulation with two (equi-amplitude)-component signals. Characteristic frequency was estimated by on-line analysis of the fiber's synchrony response pattern to a two-component signal whose bandwidth was approximately 10%–20% of the unit's CF. Estimates of fiber CF remained relatively constant over a 60 to 80 dB range of intensities. Frequency selectivity (for a two-tone complex) was determined by setting either the upper or the lower signal component equal to the unit CF and varying the frequency of the second component over an octave range. Two-component signals provide a considerably higher estimate of a fiber's frequency selectivity than is commonly observed using single-component stimuli. [Research supported by NIH.]

10:20

E6. Behavior of the pulse-number distribution for the neural spike train in the cat's auditory nerve. Malvin Carl Teich and Shyam M. Khanna (Departments of Electrical Engineering and Otolaryngology, Columbia University, New York, NY 10027)

Pulse-number distributions (PNDs) were recorded from primary afferent fibers in the auditory nerve of the cat, using standard microelectrode techniques. Pure-tone and broadband-noise stimuli were used. The number of neural spikes (pulses) n was measured in a set of contiguous intervals, each of duration T seconds. The quantity n varies from one interval to another. These data were then used to determine the PND, which is the probability $p(n, T)$ of occurrence of n spikes in the time T , versus the number n . The estimated mean and variance of $p(n, T)$ were obtained. Two different values of T were used. We observe that the count mean-to-variance ratio R is relatively constant and independent of the stimulus intensity. The PND readily exhibits the existence of spike pairs in the underlying point process for some units. A study of the scaled and unscaled pulse-interval distributions (PIDs), under conditions of spontaneous firing, demonstrates that the occurrences of neural events are sometimes not describable by a renewal process. [Work supported by NIH Grant 2-R01-NS 03654.]

10:35

E7. Acoustic response properties of units in dorsal cochlear nucleus of unanesthetized, decerebrate gerbil. Herbert F. Voigt (Departments of Biomedical Engineering and Otolaryngology, Boston University, 110 Cummington Street, Boston, MA 02215)

Single unit recordings were made in the dorsal cochlear nucleus (DCN) of unanesthetized, decerebrate Mongolian gerbils. Access to the gerbil DCN was obtained using a newly described approach [Frisina *et al.*, *Hearing Res.* 6, 259 (1982)]. Single units, recorded with 3M NaCl filled micropipette electrodes (20 Mohm typical), were classified into three types (II, III, or IV) based on their responses to best frequency (BF) tone bursts and their rates of spontaneous discharges. Type II units have little or no spontaneous activity and are excited by BF tones at all levels above threshold. Type III units are also excited by BF tones at all levels above threshold, but have spontaneous activity rates that exceed 2.5 spikes/s. Type IV units are excited over a narrow range of sound levels near BF threshold, but are strongly inhibited at higher levels. Thus all unit types found in the unanesthetized, decerebrate cat DCN [Young and Brownell, *J. Neurophysiol.* 39, 282 (1976); Young and Voigt, *Hearing Res.* 6, 153 (1982)] are also found in gerbil. [Work supported by NIH.]

10:50

E8. Temporal-neural-response correlates of pitch-intensity effects and diplacusis. K. Jones, A. Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907), and E. M. Burns (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

Pure-tone pitch anomalies such as pitch-intensity effects and diplacusis have been traditionally considered as evidence against temporal theories of pitch. These phenomena, however, may at least in part, be correlated with stochastic model calculations of VIIIth nerve firing probabilities. These calculations predict shifts in the peaks of the ISI histogram as a function of the intensity of acoustic stimulation which are consistent with average human pitch-intensity data. The same models suggest a temporal basis for some features of central-processor-type algorithms in complex-tone pitch perception. They also account qualitatively for the frequency-dependent deviation from unity of the ratio of the first ISI histogram peak to the stimulus period, as seen in ISI measurements. Physiological and psychophysical measurements, and detailed neural model studies which are designed to check the validity of these preliminary results, will be described. [Work supported by NIH Grant NS 15405 and NSF Grant BNS 8021529.]

11:05

E9. Implications of ear canal geometry for various acoustical measurements. Michael R. Stinson (Division of Physics, National Research Council, Ottawa, Canada K1A 0R6)

Recent work has demonstrated that the geometry (size, shape) of the human ear canal leads to a sound field, at high frequencies, that varies within the canal in a complicated fashion. In particular, the eardrum terminates the ear canal at a rather sharp angle, and significant variations of sound pressure (over 15 dB) arise over the eardrum surface above 10 kHz. The implications of these observations, as they apply to several aspects of hearing research, will be discussed, utilizing a three-dimensional horn equation approach that has been developed. Knowledge of the sound pressure distribution within the ear canal is important in the extension of audiometry to high frequencies, and for guiding the development of hearing aids. Studies of auditory processes that use a reference microphone in the ear canal become increasingly sensitive to microphone location as higher frequencies are used; estimates of the errors of reproducibility are given. Present acoustic network models of the middle ear assume a uniform sound pressure at the eardrum; possible ways of handling a non-uniform pressure distribution are discussed.

11:20

E10. Electrical stimulation of the auditory nerve: Membrane models applied to the interpretation of electrophysiological and psychophysical responses. M. White (Department of Otolaryngology, UCSF, San Francisco, CA 94143)

Electrophysiological and psychophysical measures of threshold as a function of sinusoidal stimulus frequency deviate from those predicted by Hill's excitation model. Several membrane models (e.g., the Frankenhauser-Huxley model and the Hodgkin-Huxley model) suggest possible mechanisms responsible for the deviations. For example, by using a modified Hodgkin-Huxley model more accurate threshold "predictions" were obtained over the 50-500-Hz frequency range. The Hodgkin-Huxley model was modified by increasing the sodium inactivation rate constant, βh , by a factor of 4. This modification caused the hyperpolarizing phase of the stimulus to become more effective at increasing the simulated membrane's excitability to the subsequent depolarizing phase, particularly when the hyperpolarizing phase was on the order of 10 ms in duration. Consistent with this model, recordings in the AVCN of cat indicate that a hyperpolarizing electrical stimulus applied to the auditory nerve significantly increases the excitability of the nerve to a depolarizing stimulus that follows. [Work supported by NIH.]

11:35

E11. Directional sensitivity of auditory neurons in the superior colliculus of the bat (*Myotis lucifugus*). Donald Wong (Department of Biology, Washington University, St. Louis, MO 63130)

The directional sensitivity of auditory neurons in the deep layers of the superior colliculus was examined in unanesthetized bats stimulated in the free field with frequency-modulated (FM) sounds (100 to 40 kHz) mimicking their natural orientation sounds. The minimum threshold of a unit was first measured when the sound was presented directly in front of the animal. Sounds 10, 30, and 50 dB above threshold were then delivered at every 10° azimuth from a hoop-mounted speaker that rotated along the horizontal plane from 80° ipsilateral to 80° contralateral. Many units exhibited spatial selectivity to a sound source that was restricted between 10° ipsilateral to 20° contralateral and had an intensity 10 dB above threshold. Sound of higher intensity was effective throughout the contralateral side in driving these neurons, although the relative response magnitude differed at the individual angles. Mapping experiments did not reveal any systematic representation of best azimuth in this midbrain structure. Nevertheless, faint echoes returning from a target along a line near the bat's flight path provide directional cues necessary for accurate target localization. [Work supported by NIH.]

11:50

E12. Susceptibility to intense impulses. G. Richard Price (U. S. Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005) and David J. Lim (Department of Otolaryngology, Ohio State University College of Medicine, Columbus, OH 43210)

Price has advanced the view that susceptibility to intense impulses of the type produced by weapons is a function of a spectrally related "critical level" in the ear [J. Acoust. Soc. Am. 73, 556-566 (1983); J. Acoust. Soc. Am. Suppl. 1 62, S95 (1977)]. This view results in a major disagreement with all damage risk criteria in use in the world by predicting that for equal peak pressures, rifle impulses are more hazardous than cannon impulses. A crucial experiment was run using both ears of 57 cats to test this hypothesis [G. R. Price, J. Acoust. Soc. Am. Suppl. 1 71, S79 (1982)]. Two months post exposure, final electrophysiological measures were made and the ears were perfused for histological processing. Histological measures involving both scanning electron microscopy and serial sectioning and light microscopy (opposite ears of each cat) can now be compared with the earlier electrophysiological measures. The two sets of histological measures show the same picture and they in turn correlate well with the electrophysiological measures.

Session F. Psychological Acoustics I: Pattern Processing in Space and Time

Lois L. Elliott, Chairman

Northwestern University, 2299 Sheridan Road, Evanston, Illinois 60201

Chairman's Introduction—9:00

Contributed Papers

9:05

F1. Behavioral evidence for central coding of azimuth as a function of stimulus frequency. Alan D. Musicant¹ (Departments of Behavioral Science and Surgery, University of Chicago, Chicago, IL 60637)

In a previous report to the Society [Musicant and Butler, J. Acoust. Soc. Am. Suppl. 1 72, S93 (1982)] we presented data on the orderly progression of perceived azimuth of narrow bands of noise as a function of stimulus frequency. The patterns that resulted have been called Spatial Referent Maps (SRMs). A recent experiment suggests that these SRMs must involve a central mechanism for coding location in the horizontal plane. In this experiment, stimuli consisted of 1-kHz-wide noise bands with center frequency (CF) ranging from 4–14 kHz. There were 13 loudspeakers placed 15° apart at locations from 360° to 180° azimuth. Stimuli were presented from only that loudspeaker located at 270°. Subjects, listening monaurally, responded with the loudspeaker position from which they perceived the sound as originating. SRMs were then constructed. Subjects next performed the same task, but now listened monaurally with the pinna cavity of the open ear filled. The external meatus was open. Azimuthal judgements of stimulus location were collected and compared to the data collected earlier. Correlations between the means of the azimuthal judgements at each frequency for the two conditions ranged from 0.81–0.97 for seven subjects. The high degree of correlation suggests that, at least for narrow bands of noise, a central mechanism must exist that codes azimuthal sound source location as a function of stimulus frequency. ¹Present address: Department of Neurophysiology, University of Wisconsin, Madison, WI 53706.

9:20

F2. Sidedness and perception for single echo of some ordinary sounds. Terry S. Zaccane and Earl D. Schubert (Hearing and Speech Science, Stanford University, Stanford, CA 94305)

Nine different sound-sample sequences ranging in temporal predictability from white noise to English speech were recorded with a single echo delayed from zero to 100 ms. The sounds were recorded in the presentation modes of monaural, dichotic, and mixed (original to both ears, delayed in one). The single repetition intensity was equal to, and 3 and 6 dB greater than, that of the original sound. Subjects were asked to indicate when the echo was perceived and, in separate tests, on which side they perceived the sound. In general, the ability of the subjects to choose the side of the leading signal diminished as the repetition delay increased past 20 ms. They were expected to perceive temporal order rather than sidedness as the delay became longer than 20 ms. Even for sounds with pronounced envelope, the subjects were not able to regain identification of sidedness out to 100-ms delay. Presentation of different sound types resulted in significant differences in performance. White noise presented the most difficult task, as shown by the low percentage of identification of the leading ear. The female singer and violin yielded the highest percentages for leading-ear identification.

9:35

F3. Pitch and spectral estimation of speech based on auditory synchrony model. Stephanie Seneff (Department of Electrical Engineering & Computer Science, Rm. 36-521, Massachusetts Institute of Technology, Cambridge, MA 02139)

This paper describes a system for processing sonorant regions of speech, motivated by knowledge of the human auditory system. The spectral representation is intended to reflect a proposed model for human auditory processing of speech, which takes advantage of synchrony in the nerve firing patterns to enhance formant peaks. The auditory model is also applied to pitch extraction, and thus a temporal pitch processor is envisioned. The spectrum is derived from the outputs of a set of linear filters with critical bandwidths. Saturation and adaptation are incorporated for each filter independently. Each "spectral" coefficient is determined by weighting the amplitude response at that frequency (corresponding to mean firing rate) by a measure of synchrony to the center frequency of the filter. Pitch is derived from a waveform generated by adding the (weighted) rectified filter outputs across the frequency dimension. The system performance is evaluated by processing of a variety of signals, including natural and synthetic speech, and results are compared with other processing methods and with known psychoacoustical data from these types of stimuli. [Work supported in part by NINCDS and the System Development Foundation.]

9:50

F4. Effect of amplitude modulation upon fusion of spectral components. Albert S. Bregman, Jack Abramson (Department of Psychology, McGill University, 1205 Doctor Penfield Avenue, Montreal, Quebec, Canada H3A 1B1), and Christopher Darwin (Laboratory of Experimental Psychology, Sussex University, Brighton, England)

We studied the perceptual integration of two complex tones, each sent to both ears and formed by amplitude modulation (AM) of a carrier frequency CF by a sinusoid with modulation frequency MF. One tone always had CF = 1500 Hz and MF = 100 Hz. The other had CFs around 500 Hz and MFs around 100 Hz. Both harmonic and inharmonic partials, produced by AM, were employed. The method involved studying the competition between two organizations: (a) the fusion of the two tones, and (b) the tendency of the higher to be stripped out of the mixture by a competing sequential organization. Fusion was best when the both tones had the same MF, even when the resulting partials did not form part of the same harmonic series. Fusion was enhanced when AM applied to the two tones was in phase. Results relate to the perceptual separation of simultaneous voices and favor a theory in which basilar membrane outputs that are amplitude modulated by the same glottal pulse will be allocated to the same voice.

10:05

F5. Modulation transfer function using temporal-probe tones. Christopher Ahlstrom and Larry E. Humes (Division of Hearing and Speech, Vanderbilt University, Nashville, TN 37212)

Psychoacoustic MTFs were measured by embedding a probe tone in SAM speech noise. The speech noise was 70 dB SPL overall, and the modulation depth of this noise was 60 dB. The modulation frequencies were: 0, 2.5, 5, 10, 20, and 35 Hz. The duration of each tone pip was 4.7 ms for frequencies 500, 1414, and 4000 Hz. Bandwidths (3 dB) at each of the frequencies were 19, 22, and 18 Hz. Skirt slopes were between 6 and 7 dB/octave. Simple masking nearly predicted changes in these MTFs in noise. A Speech Transmission Index (STI) derived from the MTFs in noise was no better correlated ($r = 0.95$) with speech recognition scores than was the

STI calculated according to H. J. M. Steeneken and T. Houtgast, J. Acoust. Soc. Am. 67, 318-326 (1980). [Work supported by the Veteran's Administration.]

10:20

F6. Temporal effects of vowel production on listeners' perception of speaker age. Herbert J. Oyer and Michael D. Trudeau (Speech and Hearing Science Section, The Ohio State University, 154 N. Oval Mall, Columbus, OH 43210)

Twenty-four speakers prolonged the vowels /a e i o u/ and read the passage "My Grandfather." The speakers were divided into four groups of six (three males, three females) based on chronological age, 41-50 years, 51-60 years, 61-70 years, and 71 years and over. Naive listeners provided perceptual judgements of each speaker's age based on the first two sentences from "My Grandfather." The speakers' vowel productions were measured to determine time required to initiate and terminate each vowel. These ten measures plus time to articulate the four syllables /maɪ grændfɑðə/ and speaker sex were regressed against speaker chronological and perceived ages. The best significant model for chronological age yielded an *R*-Square of only 0.428; while for the best significant model for perceived age *R*-Square was 0.647. Results indicate that speaker temporal control of speech exerts a strong influence on listener perception of speaker age, but has a relatively weaker relationship with speaker's chronological age. These findings suggest that temporal aspects of speech are critical in estimation of speaker age.

10:35

F7. Octave equivalence in the processing of tonal sequences. Diana Deutsch (Center for Human Information Processing, University of California, San Diego, La Jolla, CA 92093)

An issue of considerable debate concerns octave equivalence of tones that are presented in a sequential setting. In a previous study, Deutsch [Perception Psychophys. 6, 411-412 (1972)] showed that a well-known melody became unrecognizable when its component tones were displaced randomly to different octaves. It was concluded that, in accordance with the two-channel model for the abstraction of pitch relationships proposed by Deutsch [Psychol. Rev. 76, 300-307 (1969)], octave equivalence effects do not operate directly in a sequential setting. It has, however, been argued that distortion of melodic contour was responsible for the recognition performance obtained. To settle this issue, a new paradigm was employed. Musically trained listeners were presented with novel sequences of tones, which they recorded in musical notation. When the tones within a sequence were in the same octave, a high performance level was obtained. However, when the tones within a sequence were distributed across octaves, performance deteriorated sharply. This result confirms the hypothesis that octave equivalence effects do not operate directly in a sequential setting. [Work supported by NIMH.]

10:50

F8. Stapedius reflex: On some fundamental temporal characteristics. Jozef J. Zwislocki (Institute for Sensory Research, Syracuse University, Syracuse, NY 13210)

Effects of neural temporal summation and adaptation on the stapedius reflex are examined with the help of monotic and dichotic pairs of tone bursts. The tone intensities were so chosen that each burst presented separately evoked a constant response. The results show that the apparent summation of the effects of dichotic bursts is greater than that of monotic bursts and decays monotonically as the interburst time interval increases.

With monotic bursts the apparent summation effect has a flat maximum around a time interval between the burst onsets of about 60 ms. The dichotic as well as the monotic reflex arc contains at least one nonlinearity since the summated effect of the bursts exceeds the arithmetic sum. The measured characteristics can be accounted for quantitatively by a model containing a peripheral linear adaptation stage with a recovery time constant on the order of 50 ms, a linear central temporal integrator with a time constant of about 200 ms and a threshold, all connected in series. All the numerical constants are consistent with those found in neurophysiological and psychophysical experiments. The inference that the threshold follows the integrator may prove to be of major interest.

11:05

F9. A demonstration of peripheral constraint on an auditory duration discrimination task. Gregory J. Fleet and Brian R. Shelton (Department of Psychology, University of Western Ontario, London, Ontario, Canada N6A 5C2)

An earlier study investigated the discrimination of auditory durations presented at a rapid rate in a two-alternative forced-choice task. The results indicated that a constant Weber fraction was obtained across base durations from 25-1600 ms if a long interstimulus interval (ISI) separated the two alternative presentations, but performance degraded with short base durations if a short ISI was used [B. R. Shelton, J. Acoust. Soc. Am. Suppl. 1 72, S89 (1982)]. Here, the same measurements were taken under two presentation conditions: (1) a binaural presentation of both intervals, and (2) a monaural presentation of the two intervals, each to a different ear. Base durations of 25, 50, 100, and 200 ms were used, with ISI values of 25, 50, 100, 200, 400, 800, and 1600 ms. The binaurally presented condition replicated the previous results, but the presentation of intervals to alternate ears produced a marked improvement in performance, especially with short ISI presentations. This finding suggests that the discrimination of the duration of rapidly presented auditory stimuli is restricted peripherally. [Work supported by NSERC.]

11:20

F10. Auditory time constants unified. Pierre Divenyi (Speech and Hearing Research, VA Medical Center, Martinez, CA 94553 and INRS-Telecommunications, University of Quebec, Verdun, Quebec, Canada H3E 1H6) and Robert V. Shannon (Coleman Lab, 863 HSE, UCSF, San Francisco, CA 94143)

Psychophysical measures of auditory temporal integration typically yield several different time constants depending on the task. An integration time constant of 100 to 300 ms is proposed to explain the relationship between threshold and burst duration [R. Plomp and M. A. Bouman, J. Acoust. Soc. Am. 31, 749-758 (1959)]. When click thresholds are measured in amplitude-modulated noise the results are consistent with a much shorter integration time of 10 ms or less [N. F. Viemeister, in *Psychophysics and Physiology of Hearing*, edited by Evans and Wilson (Academic, New York, 1977, pp. 419-427)]. The difference between these time constants is thought to be due simply to the different tasks involved. The longer time constant is calculated assuming a linear integrator operating on the input intensity. Any physiological integration mechanism, however, integrates neural activity, which is a compressive nonlinear function of the input intensity. An integrator with a 10-ms time constant preceded by a compressive nonlinearity will produce apparent threshold integration times of 100-300 ms, depending on the magnitude of the compression. Thus a single integration mechanism can account for the apparently long time constants for threshold and the shorter time constants for temporal discrimination.

Session G. Physical Acoustics I: Novel Applications of Macrosonics

Robert E. Apfel, Chairman

*Department of Mechanical Engineering, Yale University, New Haven, Connecticut 06511***Chairman's Introduction—8:30*****Invited Papers*****8:35****G1. Another round with nonlinear acoustics.** Robert T. Beyer (Physics Department, Brown University, Providence, RI 02912)

Early work in nonlinear acoustics largely followed the behavior of one or two idealized beams of high intensity sound. More recent work has centered on attempts to make the beams more realistic in nature, and to follow up on possible ways in which nonlinear acoustics can be applied to other problems of acoustics. In the former case, such analyses as that of Zabolotskaya-Khokhlov on a tapered beam, or the use of truncated equations, especially in the study of nonlinear interaction of sound with air bubbles in water come to mind. There has also been considerable attention to the physical meaning of the nonlinear parameter. On the experimental side, much work has been done on the sound-bubble interaction. In addition, the existence of higher harmonics in a search beam of finite amplitude means that some of the limitations presented by the frequency of the fundamental beam can be favorably modified. Examples will be discussed.

9:00**G2. New applications of nonlinear acoustics at microwave frequencies.** Daniel Rugar (Edward L. Ginzton Laboratory, Stanford University, Stanford, CA 94305)

Three important new applications have been found for nonlinear acoustics in the microwave frequency regime. For the first example, the gigahertz frequency acoustic microscope is seen to significantly increase in resolution when operated at high transmitted power levels. This effect is believed to be due to energy flow from the second and higher harmonics back to the fundamental frequency after passage through the focal region of the beam. Using this effect, resolution better than $0.2 \mu\text{m}$ has been demonstrated in water at 4.3 GHz [B. Hadimioglu (to be published)]. The second application of nonlinear acoustics uses the focused beam of a liquid helium acoustic microscope to detect incoherent phonons (10 GHz) generated by a heater [J. S. Foster, D. Rugar, and C. F. Quate (to be published)]. This technique, which relies on scattering of sound by sound, has resulted in a new form of acoustic imaging. The third new application has provided the first accurate measurements of low-energy phonon dispersion in liquid helium at temperatures less than 0.1 K. For plane waves in a dispersive, lossless fluid, power in the second harmonic will periodically rise and fall as a function of propagation distance. By measuring the periodicity, dispersion can be accurately determined. This has recently been accomplished at a fundamental frequency of 2 GHz. [Work supported by ONR.]

9:25**G3. Discovery of a solitary, localized, stationary water wave.** Junru Wu, Robert Keolian, and Isadore Rudnick (Department of Physics, University of California, Los Angeles, CA 90024)

When a trough, $15 \times 1 \text{ in.}$, is filled with water to a depth of 2 cm and driven parametrically by oscillating it vertically with an amplitude of about 0.06 cm at a frequency, 10 Hz, where the half-wavelength of the surface wave is equal to the width of the trough, a striking phenomenon is observed. In a strongly localized region, which may be centered almost anywhere along the trough, a standing wave of very high amplitude is seen to oscillate across the width of the channel. Surprisingly, its amplitude drops off exponentially at its edges with a $1/e$ length of only 0.9 cm. It oscillates at subharmonic frequencies. This energy packet has particle-like behavior. When the trough is tilted the wave packet moves toward the shallow end. Two solitary waves of opposite phase, pair. Those of like phase combine or form an oscillating pair. [Work supported by ONR.]

9:50**G4. Metal fatigue testing at ultrasonic frequencies, history and uses.** Warren P. Mason (4449 Meandering Way, Tallahassee, FL 32308)

Ultrasonic frequencies in the range of 20 kHz are now being used to test fatigue in materials on account of the rapidity with which the tests can be made. It appears that the original work was done by the speaker as described in the paper. At a recent international conference "Fatigue and Corrosion Fatigue up to Ultrasonic Frequencies," which was dedicated to me on account of my original work in the field, it is stated that as a technique ultrasonic fatigue is almost as sophisticated as push-pull fatigue testing. It has been shown that the

results obtained at high frequencies are closely similar to results obtained at low frequencies, cycle per cycle. Constant total strain and constant displacement tests are possible. Positive mean load, elevated temperatures, and corrosion ultrasonic fatigue tests are now performed routinely. The feasibility and advantages of performing crack propagation tests at ultrasonic frequencies has been demonstrated. Hence ultrasonic fatigue testing is a very widely used technique.

10:15

G5. Space applications of macrosonics. Taylor G. Wang (Jet Propulsion Laboratory, 4800 Oak Grove Drive, Pasadena, CA 91109)

(Abstract not available.)

10:40

G6. Recent trends in practical applications of macrosonics. E. A. Neppiras (17 Kinsbourne Avenue, Bournemouth, England)

Macrosonics includes the area of technology where treatment by sound or ultrasound produces permanent and useful changes in the treated medium. To date, two types of application have achieved major importance as viable commercial techniques. These—cleaning and plastics welding—are still expanding in scope and importance. Recent progress has been mainly in the direction of a greater degree of automation and control. Apart from these two established areas there are several applications that have already been more-or-less fully developed but which, for one reason or another, have a much more limited scope. Examples are: ultrasonic machining, metal-welding, tinning, and various metal deformation processes. A third category consists of applications and ideas that appear very promising but which have not yet been fully explored or developed. Examples are: some aspects of chemical processing and various gas-load applications, including the production of aerosols (atomization). After briefly reviewing the latest progress in the established areas, this paper will concentrate on new ideas and applications that show signs of developing into valuable practical techniques.

11:05

G7. Industrial machines employing resonance wave systems that are inductively excited by reactive mass vibrators. A. G. Bodine (Bodine Soundrive Company, 7877 Woodley Avenue, Van Nuys, CA 91406)

For the field of macrosonics employing resonance technology with a mechanical oscillator, the system is driven at a low impedance region in order to have ample current flow (elastic stroke) at the oscillator. The oscillator is analogous to a constant voltage generation, so it has to function with an unconstrained or variable amplitude rather than with a fixed stroke limitation like a crank mechanism, for instance. The true mechanical oscillator is thus an inductive coupling of a dynamic mass. The best form is an orbiting mass describing its path in free space except for the constraint which confines the mass to its orbit. It is the inductively phased reactance of this constraint which provides the constant voltage ac drive for the resonance system frequency. This kind of oscillator thus adapts its cyclic stroke to the freedom of the resonance system. A very useful technique is to determine and maintain the frequency of the oscillator at that which would be the resonance frequency of the system if the work load at the output region of the system were infinite. Maximum Q thus occurs with a "locked" load. Then, for all finite loads, the output amplitude stays constant because a drop in load means reduction in mechanical amplification as the fixed frequency system strays from the peak resonance wave pattern environment.

Contributed Paper

11:30

G8. TDS measurement of the second harmonic emission from ensonified bubbles. Richard C. Heyser and James A. Rooney (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

The second harmonic spectrum of backscattered sound from air bubbles in water has been measured by TDS methods. Using the technique of Miller and Nyborg [D. L. Miller and W. L. Nyborg, *J. Acoust. Soc. Am.* 73, 1537-1544], air was trapped in the pores of immersed hydrophobic Nuclepore filter membranes to produce stable bubbles whose diameters lay in the 3-4 μ range. The membrane was insonified with a wideband transmitting crystal driven from a linear amplifier, and the backscattered sound was received by a second wideband ultrasound crystal. Transmis-

sion in the 0.5- to 10-MHz range was accomplished by a digitally controlled sweeping generator operating at a rate of 500 MHz/s. Second harmonic reflection of energy was detected by a tracking receiver whose sweep program was driven synchronously with the transmitting program, but with a tracking rate of 1 GHz/s and a delay offset corresponding to the arrival time of backscattered sound from the membrane. Selected combinations of transmission and reception sweep rate established that the observed second harmonic signal was due to resonance of bubbles and not due to incidental transmitter or receiver distortion components. The measured frequency spectrum of second harmonic backscatter from bubbles corresponds closely to the predicted spectrum shape, showing a dominant peak at twice the frequency at which a strong absorption dip occurs for sound passing through the bubble-filled membrane.

Session H. Shock and Vibration I: Instrumentation

Ronald L. Bannister, Chairman

Westinghouse Electric Corporation, Steam Turbine Division, Orlando, Florida 32807

Chairman's Introduction—9:00

Invited Papers

9:05

H1. An overview of shock and vibration measurements. Rudolph H. Volin (Shock and Vibration Information Center, U. S. Naval Research Laboratory, Washington, DC 20375)

Early instruments limited our ability to make shock and vibration measurements; however, the development of new instruments and measuring techniques has advanced the state of the art in shock and vibration measurement. The most significant developments were the piezoelectric accelerometer and the Fast Fourier Transform algorithm. Both of these developments, coupled with advances in the digital computer technology, significantly influenced the current state of the shock and vibration measurement technology. Piezoelectric accelerometers enable us to measure complex motions of lighter structures to higher frequencies. The Fast Fourier Transform algorithm initially made it possible to analyze shock and vibration test data on main frame digital computers rapidly and economically; but, combined with advances in the digital computer technology, it has broadened our ability to conduct many types of laboratory shock and vibration tests, make measurements, and analyze test data. Advances in the digital computer technology will continue to influence the progress in shock and vibration measurement. For example, the microcomputer has already proven to be effective in certain data analysis applications. Future increases in their computing power will allow them to be used in a wider range of measurement and data analyses applications.

9:30

H2. Array processing to eliminate standing wave effects in shell radiation measurements. E. V. Thomas (DTNSRDC, Code 2742, Annapolis, MD 21402)

The spectral measurement of a cylindrical shell's radiation characteristics in a tidal basin were desired to compare with deep ocean farfield radiation characteristics. Standing wave effects from the surface and bottom of the basin spectrally distorted the radiation measurements. The techniques evaluated to provide relief from standing wave effects were: large, highly directive arrays; line arrays in the nearfield with detection before averaging; and line arrays in the semifarfield with and without beamforming. Detection methods were linear, square law, and logarithmic. An analytical evaluation of the detection methods indicated linear or logarithmic detection had a lower standard deviation than square-law detection. Also, a large, highly directive array wrapped around the shell (i.e., the very nearfield) was evaluated. The best smoothing of the standing wave effects of the medium were achieved using a ten-element line array (normal to cylinder axis) in the semifarfield using logarithmic detection before averaging. Further experimental work simplified this array to a three-element sloped line with logarithmic detection. This system allowed measurement of the shell's radiation characteristics over 2½ decades without perturbation by standing waves.

9:55

H3. Underwater modal analysis of heat exchanger tubing using pressure transducers. C. D. Powell, C. P. Dietz, and N. J. Middelburg (O'Donnell & Associates, Inc., 241 Curry Hollow Road, Pittsburgh, PA 15236)

In order to perform an accurate flow-induced vibration analysis of heat exchanger tubing, natural frequencies, mode shapes, and damping values must be known. This study concerns experimental determination of the above parameters for titanium tubing in an underwater environment. Being a lightweight material with high stiffness, it was anticipated that high natural frequencies with small damping values would be encountered which could be erroneously altered by the mounting and cabling of traditional instrumentation such as accelerometers. A 1-g microaccelerometer was used to perform a preliminary modal survey to determine system natural frequencies, mode shapes, and separation of tube and fixture modes. It was found that mounted instrumentation did indeed cause mass loading problems resulting in erroneous natural frequencies and damping values. Noncontacting pressure transducers were used to confirm the identical tube mode shapes and to then determine true tube natural frequencies and damping values using a variety of excitation techniques. The study concludes by evaluating the most applicable techniques to be used in order to obtain valid data.

10:30

H4. Application of advanced excitation and analysis techniques to the Galileo Spacecraft modal survey program. David L. Hunt (SDRC Inc., 11055 Roselle Street, San Diego, CA 92121)

A modal test of the Galileo Spacecraft will be used to tune a finite element model that will serve in a subsequent loads analysis. A part of the test is being devoted to applying new techniques for performing modal tests. Testing efforts will be applied in the areas of multi-shaker broadband (MSB) excitation, time domain calculation of rational frequency response functions, and polyreference estimation of modal parameters. These three techniques, having been developed and implemented, are being examined as a means to perform modal tests in a fraction of the time previously required, with a resulting data quality which meets or exceeds current standards. The presentation will detail the techniques, their application to the specific Galileo program, and highlight and interpret the results achieved.

10:45

H5. Using holographic interferometry in vibration measurements. R. Page (Apollo Lasers, Inc., 9201 Independence Avenue, Chatsworth, CA 91311)

This paper will review principles of holographic interferometry and provide examples of current applications in vibration analysis. Specific techniques discussed will include double exposure (frozen fringe), real time, and stroboscopic and time-average holographic interferometry.

Contributed Papers

11:10

H6. The NBS conical transducer. T. M. Proctor, Jr. (National Engineering Laboratory, National Bureau of Standards, Washington, DC 20234)

The NBS conical transducer was first reported on in 1980 [T. M. Proctor, Jr., *J. Acoust. Soc. Am. Suppl.* 1 68, S68 (1980)]. A number of changes have been made in this transducer that improve its usefulness and its frequency response for acoustic emission work. A design for use as a secondary standard in the calibration of other acoustic emission transducers is discussed. A smaller version has been constructed with virtually the same frequency response as the original version. Electrode problems are discussed and solutions indicated.

11:25

H7. Transient waves in an elastic plate. T. M. Proctor, Jr., and F. R. Breckenridge (National Engineering Laboratory, National Bureau of Standards, Washington, DC 20234), and Y-H. Pao (Cornell University, Ithaca, NY 14853)

Recently Pao *et al.* [Y-H. Pao, R. R. Gajewski, and A. N. Cevanoglu, *J. Acoust. Soc. Am.* 65, 96-105 (1979)] have obtained theoretical solutions for wave propagation in an elastic plate by using generalized ray theory. These solutions give the dynamic displacement at locations on the surface of a plate due to a point force step function input. Using the NBS conical transducer, a broadband displacement transducer, experimental verification of the motion has been obtained.

TUESDAY MORNING, 8 NOVEMBER 1983

TOWN AND COUNTRY ROOM, 9:00 A.M. to 12:31 P.M.

Session I. Speech Communication I: Speech Recognition Algorithms

Janet M. Baker, Chairman

Dragon Systems, Inc., 173 Highland Street, West Newton, Massachusetts 02165

Chairman's Introduction-9:00

Contributed Papers

9:05

I1. An acoustic analysis of coarticulatory phenomena in connected digit strings. Marcia A. Bush (Fairchild Laboratory for Artificial Intelligence Research, 4001 Miranda Avenue, MS 30-888, Palo Alto, CA 94304), Marie Hamilton, and Kazue Hata (Department of Linguistics, University of California at Berkeley, Berkeley, CA 94720)

To date, relatively few attempts have been made to explicitly incorporate coarticulatory information in the design of automatic systems for connected digit recognition. One reason for the limited progress in this area has been the lack of a convenient description of the relevant coarticulatory phenomena, culled from systematic analyses of large amounts of real speech data. This paper describes the initial stage of an ongoing project aimed at remedying this situation. The paper is based on computer-

aided analyses of waveforms and spectrograms of 1180 isolated digits and connected digit strings, as produced by two male and two female speakers of American English. The acoustic data were supplemented with phonetic transcriptions by trained linguists. Coarticulatory phenomena are categorized according to: (1) word-boundary effects (e.g., phoneme insertions and deletions); (2) within-word effects (e.g., context-dependent changes in vowel formant frequencies); and (3) consistent speaker-dependent effects (e.g., allophonic variations of /r/). Implications of the acoustic data for system design are also discussed. In particular, the view is advanced that certain coarticulatory phenomena may serve as important sources of information in identifying "difficult" digit sequences (e.g., those containing the digit "oh").

9:17

12. Exploring phonotactic lexical constraints in word recognition. Daniel P. Huttenlocher and Victor W. Zue (Room 36-549, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

In a previous meeting of the Society, Zue and Shipman demonstrated that the constraints imposed by the allowable sound sequences of a language are extremely powerful. Even at a broad phonetic level of representation, sequential constraints severely limit the number of possible word candidates [J. Acoust. Soc. Am. Suppl. 1 71, S7 (1982)]. This provides an attractive model for lexical access based on partial phonetic information. However, Zue and Shipman's results did not take into account the fact that the acoustic realizations of phonetic segments are highly variable, and this variability introduces a good deal of recognition ambiguity in the initial classification of the signal. We have conducted a set of studies investigating the robustness of sequential phonetic constraints with respect to variability and error in broad phonetic classification. In these studies segment misclassifications or deletions are permitted. In one study it was found that the phonetically variable parts of words (around reduced syllables) provide much less lexical constraint than the phonetically invariant parts. Thus, by utilizing the information from robust parts of a word, a large lexicon can still be partitioned into small equivalence classes. Detailed results of the studies will be presented. [Work supported by the Office of Naval Research under contract N00014-82-K-0727 and by the System Development Foundation.]

9:29

13. Exploring allophonic and lexical constraints in a continuous speech recognition system. Francine R. Chen and Victor W. Zue (Room 36-545, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

Recent research [Zue and Shipman, J. Acoust. Soc. Am. Suppl. 1 71, S7 (1982); Huttenlocher and Zue, this meeting] has shown that a broad phonetic representation of speech provides strong constraints for lexical access of *isolated* words. In addition, Church (Ph.D. thesis, MIT, 1983) has demonstrated the utility of *detailed* allophonic constraints for parsing a sentence from a phonetician's transcription. This gives reason to believe that the coupling of lexical and allophonic constraints can be a powerful tool in *continuous* speech recognition. The present study explores how broad phonetic constraints can be applied to a restricted continuous speech task. Using a broad phonetic representation derived from an ideal transcription, it was found that on the average, 70% of the word boundaries in the digit vocabulary can be identified. Extending this approach to speech data, we have implemented a classifier which derives a broad phonetic representation from the speech signal. Preliminary results indicate that allophonic and lexical constraints can be effective in reducing the number of string candidates, based upon the output from this classifier. [Work supported by the Office of Naval Research under contract N00014-82-K-0727 and by the System Development Foundation.]

9:41

14. The use of phonotactic constraints to determine word boundaries. Lori F. Lamel (Room 36-545, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

Phonotactic constraints limit the permissible word internal consonant sequences in English. In some cases, knowledge of the phoneme sequence uniquely specifies the location of the word boundary, while in other cases, phonological rules based on allowable consonant sequences are not sufficient. For example, the word boundary can be uniquely placed in the sequence /...mgl.../, as in the word pair "some glass," whereas the word boundary location is ambiguous in the phoneme sequence /...str.../ without further acoustic information. The /...str.../ may have a word boundary in one of three places as in "last rain", "race trials", and "may stretch." Studies were conducted to determine the utility of phonotactic constraints to predict word boundaries. The data bases included the Merriam-Webster Pocket Dictionary, a phonemically balanced set of sentences, and samples of unrestricted text. Results indicate that: (1) word internal consonant sequences represent a very small subset of all permissible consonant sequences across word boundaries; and (2) acoustic-phonetic knowledge is needed when the word boundary is ambiguous. Results on the differences between word and syllable boundaries will also be presented. [Work supported by the Office of Naval Research under contract N00014-82-k-0727 and the System Development Foundation.]

9:53

15. Recognition of a small vocabulary spoken by two dialect groups. R. W. Bossemeyer and M. Laferriere (American Bell, Inc., 2220 Asbury Ave., Neptune, NJ 07753)

A major difference among American English dialects lies in the variety of phonetic characteristics of vowels (H. Kurath, *Studies in Area Linguistics*, 1972). Since LPC-based speech recognition systems depend heavily on the vowel portion of an utterance, dialect differences not represented in a template inventory could affect recognition accuracy in a speaker-independent, LPC-based isolated word recognition system. Eighty-five speakers from each of two dialect groups, Eastern New England and Southeastern, recorded the digits and two command words. Word recognition scores for speakers in each dialect group were compared when reference templates were from (1) the other dialect group; (2) the same dialect group; and (3) a combination of the two groups. Comparisons are made on a word-by-word basis to determine what types of phonetic dialect variation affect recognition accuracy.

10:05

16. An improved word detection algorithm for telephone quality speech. J. G. Wilpon, L. R. Rabiner, and T. Martin (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

Accurate location of the endpoints of spoken words and phrases is important for reliable and robust speech recognition. The endpoint detection problem is fairly straightforward for high-level speech signals in low-level stationary noise environments (e.g., signal-to-noise ratios greater than 30 dB). However, this problem becomes considerably more difficult when either the speech signals are too low in level (relative to the background noise), or when the background noise becomes highly nonstationary. Such conditions are often encountered in the switched telephone network when the limitation on using local dialed-up lines are removed. In such cases the background noise is often highly variable in both level and spectral content due to transmission line distortions, transients, and tones from the line and/or from signal generators, etc. Conventional speech endpoint detectors have been shown to perform very poorly (on the order of 50% word detection) over these conditions. In this talk we present an improved word detection algorithm which incorporates both vocabulary and task information, leading to word detection accuracies close to 100% for isolated digit detection over a wide range of telephone transmission conditions.

10:17

17. A pretty good formant tracker. Ronald A. Cole (Department of Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213) and Robert A. Brennan (Department of Electrical Engineering, Carnegie-Mellon University, Pittsburgh, PA 15213)

Algorithms were developed for estimating formant frequencies on a time frame by time frame basis. The algorithms incorporate acoustic phonetic knowledge about formants. The algorithms are based on knowledge of acoustic phonetics, and do not merely rely on spectral peaks. Thus two formants will be identified even when the first and second formants merge or when the second and third formants merge to form a single peak. The algorithms were evaluated by comparing formants drawn on a speech spectrogram by a trained phonetician to those drawn automatically. The data base consisted of both male and female speakers speaking phonetically balanced sentences. In addition, synthesis derived from the formant frequencies and amplitudes was tested for intelligibility. Results will be presented for these two procedures. [Supported by NSF and DARPA.]

10:29

18. Unsupervised speaker adaptation in feature-based isolated letter recognition. Moshé J. Lasry and Richard M. Stern (Departments of Electrical Engineering and Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213)

10:41–10:55

Break

Contributed Papers

10:55

19. On the use of hidden Markov models for speaker-independent recognition of isolated words from a medium size vocabulary. L. R. Rabiner, S. E. Levinson, and M. M. Sondhi (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

Recent work at Bell Laboratories has shown how the theories of LPC Vector Quantization (VQ) and hidden Markov modeling (HMM) can be applied to the recognition of isolated word vocabularies. Our first experiments with HMM based recognizers were restricted to a vocabulary of the ten digits. For this simple vocabulary we found that a high performance recognizer (word accuracy on the order of 97%) could be implemented, and that the performance was, for the most part, insensitive to parameters of both the Markov model and the vector quantizer. In this talk we extend our investigations to the recognition of isolated words from a medium size vocabulary, (129 words), as used in the Bell Laboratories airline reservation and information system. For this moderately complex vocabulary we have found that recognition accuracy is indeed a function of the HMM parameter (i.e., the number of states and the number of symbols in the vector quantizer). We have also found that a vector quantizer which uses energy information gives better performance than a conventional LPC shape vector quantizer of the same size (i.e., number of codebook entries).

11:07

110. Stop identification using hidden Markov models. Gary E. Kopec (Fairchild Laboratory for Artificial Intelligence Research, 4001 Miranda Avenue, MS 30-888, Palo Alto, CA 94304)

A series of experiments has been undertaken to assess the power of discrete spectral slices for automatically discriminating between the voiceless plosives /p,t,k/ in CV syllables. These experiments involve a 936-token data base consisting of 52 instances of each of the 18 syllables /p,t,k/ × /i,e,ae,a,ow,u/ spoken by 13 male and 13 female talkers. In one experiment, identification was attempted using a single 12-pole LPC onset spectrum. The onset spectra from each talker were compared, using a log-likelihood distance measure, with the 900 onset spectra of the remaining 25 talkers. An overall classification accuracy of 92% was achieved using a *k*-nearest-neighbor decision strategy. A second experiment involves classifiers which use a series of LPC spectra computed during the first 50 ms of the stop release. Each CV syllable is modeled as a hidden Markov process which generates a spectrum every 5 ms. Classification is performed using either a Viterbi or forward-backward decoding strategy.

11:19

111. A finite state parser for use in speech recognition. Kenneth W. Church (Bell Laboratories, Murray Hill, NJ 07974)

This paper describes two new methods by which a feature-based isolated letter recognition system, FEATURE [Cole *et al.*, Proc. ICASSP-1983, pp. 731–733], can learn the acoustical characteristics of individual speakers without feedback from the user. The original tuning algorithm of FEATURE made use of labeled observations from a given speaker, and of the *a priori* covariances of the feature mean values within the across decision classes, to optimally adjust the expected feature values of all the decision classes. In the first new procedure the system assumes a correct decision every time it classifies a sound with a high confidence level. The class-conditional mean vectors and covariance matrices are then computed in the same way as in the tuning-with-feedback procedure. In the second new procedure the system records the average observed feature values over all possible classes on a feature-by-feature basis. The *a priori* covariances between the mean values of a feature for each decision class with its mean values over all possible classes are then used to update the class-conditional means and covariances. Experiments were conducted on confusable sets of letters using both speaker adaptation procedures. A significant improvement in letter recognition performance using both analyses was observed after a small number of iterations. [Supported by NSF and DARPA.]

The application of finite-state parsing techniques is proposed for the purpose of assigning syllable structure to a phonetic transcription, subject to allophonic and phonotactic constraints. In this way, a speech recognition device can exploit contextually VARIANT cues in order to distinguish minimal pairs such as "a Tease/aT ease," "nighT Rate/niTRate," and "greaT Wine/gray TWine." Our parsing suggestion provides an elegant implementation framework for exploiting the important interaction between segmental variation and syllable/word context. Other examples will be discussed, including Klatt's "Did you hit it to Tom?" Finally, the framework will be extended to account for "feature spreading" (agreement and coarticulation) in a natural way, by reformulating parsing in terms of four matrix operations: addition, multiplication, transitive closure, and element-wise intersection.

11:31

112. A comparison of dynamic time warp algorithms for connected word recognition. Stephen C. Glineski (Bell Laboratories, 600 Mountain Avenue, Rm. 7C-427, Murray Hill, NJ 07974)

The application of dynamic time warping to the problem of connected word recognition has recently received much attention. Several successful approaches have appeared in the literature including the level building algorithm [C. Myers and L. Rabiner, IEEE Trans. Acoust. Speech Signal Process. ASSP-29, 284–297 (1981)] and the single pass algorithm [J. Bridle, M. Brown, and R. Chamberlain, Proc. 1982 IEEE ICASSP]. A single mathematical description of dynamic time warping is presented that unifies these and other approaches, and highlights their similarities and differences. By testing the algorithms with a database of connected digits, it is found that in general, the use of additional levels of computation results in a relatively small decrease in recognition error.

11:43

113. Reclaiming temporal information after dynamic time warping. Hollis L. Fitch (Institute for Defense Analyses, Thanet Road, Princeton, NJ 08540)

Template matching methods use dynamic time warping to eliminate differences in duration between the template and the unknown speech. These differences may be linguistically relevant. It has been demonstrated for one test word that the relative durations of certain segments in that word can be used to distinguish it from most "false alarms"—stretches of speech other than the test word that score well on the spectrally-based template match [H. L. Fitch, Proc. ICASSP 82, 1247–1250 (1982)]. Here, a more general procedure is described for segmenting, and testing the relative durations of the segments. This procedure has now been applied to ten words, and shows promising results.

11:55

I14. Vowel-space normalization procedures and phone-preserving transformations of synthetic vowels. T. M. Nearey (Department of Linguistics, University of Alberta, Edmonton, Canada T6G 2H1)

A variety of speaker-normalization procedures have been proposed for the recognition of vowels of different speakers and/or the cross-language comparison of vowel systems. Many such procedures are special cases of affine transformations of coordinates in a two-(or more) formant space. Results of perceptual experiments are presented indicating that only a limited subset of affine transformations preserve the phonetic identity of a set of synthetic vowels. From a perceptual point of view, many vowel normalization procedures appear to be too powerful in the sense that they can "over-normalize" vowels which are phonetically distinct. In particular, rotations of a two-formant space and/or independent multiplicative scaling of the F_1 and F_2 axes lead to marked change in phonetic quality. On the other hand, results indicate that uniform multiplicative scaling of F_1 and F_2 by a single constant is "almost phone-preserving" in the range of male/female vowel differences. For larger (e.g., male/child) differences, a specific nonuniform scaling similar to that proposed by Fant seems to be required to maintain "perceptual constancy."

12:07

I15. Speaker normalizing transforms for automatic recognition. Vladimir Sejnoha¹ (Department of Electrical Engineering, McGill University, Montreal, Quebec, Canada) and Paul Mermelstein (McGill University, Bell-Northern Research and INRS Telecommunications (University of Quebec), Verdun, Quebec, Canada H3E 1H6)

A method for the reduction of inter-speaker differences by spectral transformation will be described. Transforms were designed for the additive compensation and frequency scaling of speech spectra based on the log of the energies of 20 channels spaced on the mel-scale. The frequency scaling was implemented as either a linear scaling, or as a nonlinear frequency warp found by dynamic programming. The transforms were then combined so that the additive component principally addressed the differences in the tilt of the spectrum while the frequency scaling mainly treated the dissimilarities in spectrum resonance locations. The transforms were applied to a data base of average spectra of ten steady-state voiced-speech segments for seven male and six female speakers. Both the additive com-

pensation and frequency warping, and particularly the integration of the two, was useful for both within- and across-sex normalization. The overall error rate in recognition experiments on the segment data base was reduced by 50% across sex, 60% within the female speaker group, and by 20% for the male speaker group. The best frequency warp paths were found to be nonlinear and strongly dependent on the speech segment. Parameters for a single, speaker-dependent, combined transform were also derived and the global transform was observed to be as effective as the equivalent speech-segment-dependent transform. The global transform parameters derived from as few as three segments yielded a performance level close to that attained with parameters extracted from the whole set of segments for one speaker. [Research supported by NSERC, Canada.]

¹Present address: Kurzweil Alpha Systems Inc., 57 Wells Ave., Newton, MA 02159.

12:19

I16. A study of correlation methods for phoneme identification. A. Maynard Engebretson and John P. Tadlock (Central Institute for the Deaf, St. Louis, MO 63110)

With current availability of high-speed digital signal processors and inexpensive memory, correlation methods of real-time speech recognition may be practical. In the methods presented here, speech is segmented into overlapping 25.6-ms windows. A spectrum is calculated for each window and correlated with speaker-independent reference patterns. Each windowed speech fragment is identified with the reference pattern yielding the highest correlation. Average spectral patterns are created automatically from "stable" segments of speech. A high correlation between spectra of adjacent windows is used as a measure of stability. The average patterns are categorized across speakers and averaged within categories to create the reference patterns. We have studied standard and Mellin (scale invariant) correlations in combination with linear magnitude and cepstral-smoothed log magnitude spectra. Mellin correlation normalizes spectra that are scaled with respect to frequency. Cepstral-smoothing reduces the number of coefficients required to represent the spectrum. Also cepstral coefficients can be normalized with regard to transmission characteristics. Results will be presented for vowels recorded from six male and six female speakers. [Work supported in part by NIH Grant NS-03856].

TUESDAY MORNING, 8 NOVEMBER 1983

DEL MAR/HELIX/SANTA FE ROOMS, 9:00 A.M. TO 12:05 P.M.

Session J. Musical Acoustics I: Computer-Based Analysis and Synthesis of Music

Linda A. Seltzer, Co-Chairman

Department of Music, University of California, San Diego, La Jolla, California 92093

Max V. Mathews, Co-Chairman

Bell Laboratories, Murray Hill, New Jersey 07974

Chairman's Introduction—9:00

Invited Papers

9:05

J1. Digital processing of instrumental sounds. Roger Reynolds (Music Department, University of California, San Diego, La Jolla, CA 92093)

In work at IRCAM (Paris), digital processing techniques were developed to alter pre-recorded instrumental materials in perceptually novel ways. The goal of this processing was to create sound materials that were

fundamentally unfamiliar but could co-exist with natural instruments in musically satisfying ways. (1) Because sounds are represented numerically for digital purposes, the potential for sound editing is changed qualitatively. Musical phrases were extended and transformed by replicating segments of them in ordered patterns whose structural properties dominated those of the original without eclipsing its identity. (2) Phase vocoding techniques were used to analyze instrumental sounds. Resulting data were appropriately reduced to form the basis of resynthesis through the MUSIC X program. In this way, independent access to arbitrary groups of partial components was attained. Odd and even partials were presented over spatially distinct speaker channels and subjected to dynamically changing vibrato functions with differing rates and depths. Additional distinct auditory images arose in the stereo field while the sonic image of the original instrument remained. Examples of instrumental recordings and their digital transformations are presented in isolation and in musical contexts.

9:35

J2. Changing conceptions of pitch structure and timbre: A modest proposal. Gerald J. Balzano (Department of Music, University of California at San Diego, La Jolla, CA 92093)

Two areas of musical exploration of interest to composers for which the computer is unusually well-suited are microtonal systems and "new" timbres. Notions about how to achieve these musical extensions are doubly theory-dependent in that they depend on both our theory of perception and our theory of the stimulus. Most extant attempts to do microtonal music have used small-integer ratios as the fundamental description of the musical elements involved and the basic entities to which human perceivers are presumed sensitive. For timbre, the universe of possibilities has traditionally been characterized in terms of spectral variables (possibly time-varying), and more recently in terms of projections on axes in a "subjective" multidimensional space. In the present paper, alternatives to prevailing conceptions of both pitch and timbre are considered. A symmetry-oriented group-theoretic approach to pitch structure that eschews ratios will be described; the resulting view of pitch systems leads to some novel ideas about how to do microtonal music on a computer. For timbre, an approach that focuses on the dynamics of the sound-producing activity rather than on the resulting spectra will be described. This leads naturally to an interesting alternative to currently popular methods for generating timbres and "timbre transformations" on a computer.

10:05

J3. Some experiments with compositional algorithms. Charles Wuorinen (670 West End Avenue, New York, NY) and Mark Lieberman (Bell Laboratories, Murray Hill, NJ 07974)

(Abstract not received)

10:35

J4. Shakuhachi pitch and intonation: Application to computer music composition. Linda A. Seltzer (M/A-COM Linkabit Corporation, 3033 Science Park Road, San Diego, CA 92121 and Department of Music, University of California, San Diego, La Jolla, CA 92093)

The music for the Japanese bamboo flute, shakuhachi, contains microtonality as well as continuous change of pitch in the course of long notes. Shakuhachi pitch and intonation may be studied by means of pitch detection analysis of the waveform. Or, alternatively, the pitch of a note may be described as the result of the following parameters: fingering, vocal tract shape, distance of the lips from the mouthpiece, and direction of breath. The melodic technique of the shakuhachi may be applied to computer music composition in the continuous frequency domain, without restriction to a scale consisting of discrete frequencies

11:05

J5. Digital sound synthesis for underwater music perception. Michel Redolfi and Lee Ray (Computer Audio Research Laboratory, Center for Music Experiment, University of California at San Diego, La Jolla, CA 92123)

A piece of original electro-acoustic music, *Sonic Waters II*, was created. Software "instruments" and "note lists," descriptions of the behavior of those instruments in the musical time of a sequence, were specified for the *music* synthesis program to produce, by frequency modulation, amplitude modulation, and localization and motion within a synthesized space, morphological transformations of several harp recordings. Means of (1) transduction of this wide-band musical work (20 Hz to 16 kHz) for human listeners immersed in water and (2) compensating for changes of such musically significant psychoacoustic cues as depth perception and relative amplitudes of overtones occasioned by the prominence of bone conduction in underwater listening were investigated. One pressure-loaded diaphragm and four piezoelectric transducers were positioned to broadcast the computer-processed sound. The wavelength of some bass notes exceeded the dimensions of the pool, limiting musically useful propagation of low frequencies due to cancellation and reinforcement of standing waves. [Work supported by the CME, System Development Foundation and the French government.]

11:35

J6. Use of data reduction in computer music. James W. Beauchamp (2136 Music Bldg., 1114 W. Nevada, University of Illinois at Urbana-Champaign, Urbana, IL 61801)

Two competing methods of digital sound production are "instant replay" and "synthesis by algorithm." If fidelity to an original sound were the paramount consideration, replay would be the more desirable method. However, replay requires massive amounts of memory and is inflexible with respect to contextual modification, an important concept for composers. On the other hand, algorithmic synthesis methods generally are designed to minimize memory requirements and employ a few, well-chosen, perceptually significant parameters which are varied in an ad hoc fashion. Data reduction, where synthesis algorithm parameters are automatically extracted from the original sound, offers a link between instant replay and algorithmic synthesis. It is particularly useful if the extracted parameters are perceptually meaningful and therefore can be intelligently modified by the composer. Various methods of data reduction will be discussed together with their associated synthesis algorithms (including constant waveform, additive, filter, nonlinear/filter, and waveform interpolation synthesis), and sound examples will be played.

11:50

J7. Modification of musical sounds by means of the phase vocoder. Mark Dolson (Computer Audio Research Laboratory, Center for Music Experiment, Q-037, University of California, San Diego, CA 92093)

The usefulness of the phase vocoder for performing time-scale modification of speech signals is well established. However, the application of the phase vocoder to musical signals is far less common. Musical applications of the phase vocoder differ from speech applications in several important respects: (1) Musical sampling rates are much higher; hence, minimization of computation and storage requirements is far more important. (2) Musical fidelity requirements are also much higher; this counteracts attempts to minimize computation and storage. (3) Musical applications often require analysis of individual harmonics; in this case, filter spacing must match harmonic spacing. (4) Pitch translation (as opposed to time-scale modification) requires spectral envelope estimation to sound convincing. (5) Musical signals may be polyphonic. (6) Lastly, musical applications frequently seek to distort sounds in an interesting way; hence, they often involve extremes of time and pitch scaling, selective modification of analysis data, and novel cross-syntheses and timbral interpolations. Examples will be presented from a phase vocoder package designed explicitly for musical applications.

Session K. Underwater Acoustics II: Arctic Acoustics II

Henry Kutschale, Chairman
Lamont-Doherty Geological Observatory, Palisades, New York 10964

Chairman's Introduction—1:30

Invited Papers

1:35

K1. Characterization of sea ice. James P. Welsh (Head, Polar Oceanography Branch, Naval Ocean Research & Development Activity, National Space Technology Laboratories, NSTL Station, MS 39529)

Some definitions of icebergs and sea ice are presented. Various classification schemes for sea ice are discussed. The more common sea ice characteristics are presented with examples of measurement techniques and ranges of measured values. Consideration is given to potential for inference of acoustic properties from observed and measured sea ice characteristics. General overview is provided for applications of Arctic ice properties measurement experiments. These experiments include *in-situ*, airborne, and satellite measurements to obtain surface truth of remotely sensed ice characteristics particularly in the microwave frequencies.

2:00

K2. Low-frequency acoustic propagation in Canadian Arctic waters. Gary H. Brooke (Defence Research Establishment Pacific, Victoria, BC, Canada V0S 1B0)

Underwater acoustic propagation within, and surrounding, the Canadian Arctic Archipelago is typical of shallow Arctic water. That is, the propagation is strongly dependent on bottom properties at lower frequencies (< 40 Hz) but is increasingly influenced by the surface at higher frequencies due to the characteristic upward refracting velocity profile in the water column. Water depths are generally less than 600 m and are typically 200–400 m. DREP has made propagation measurements in several locations and, hence, under a variety of propagation conditions. In this paper the salient features and trends in the data are discussed.

2:25

K3. High-frequency acoustics in the Arctic. Robert E. Francois (Applied Physics Laboratory, University of Washington, Seattle, WA 98195)

Compared with other oceans, acoustic propagation in the Arctic is generally limited in range by upward refraction, ice surface interaction, and the increased absorption loss at low water temperatures. At shallow depths beneath the under-ice surface, the water column is very near its freezing point during most of the year. Considerable salinity variation occurs in the freezing season because of brine displacement as ice is formed and in the melt season because of the layering of the nearly fresh ice melt water over the sea water. This results in a positive density gradient with a rather sharp increase at 20–50-m depth, where the older Arctic water is encountered. A dense biological population, analogous to the deep scattering layer seen in the open ocean, is observed at this pycnocline. This thin layer, has high volume reverberation strength compared to values observed in other oceans. A study of these layers sometimes reveals the presence of internal waves. When propagation takes place through this layer, anomalously high absorption attributed to scattering is observed. At 60–300 kHz, we have observed apparent increases in absorption of from 10 to 20 dB/km. The presence of the ice cover decouples the wind from the water mass, resulting in greatly reduced boundary layer mixing. This allows the many discrete layers formed by the natural processes noted above to persist over large areas for long times. As a consequence large acoustic fluctuations often occur for small vertical displacements of source or receiver. [Work support by NAVSEA.]

2:50

K4. Site dependence of low-frequency Arctic ambient noise and signal attenuation. Orest I. Diachok and Stephen C. Wales (Code 5160, Naval Research Laboratory, Washington, DC 20375)

Our understanding of two significant, ice-related, low-frequency (below 1 kHz) underwater acoustic phenomena, characteristic of both Arctic and sub-Arctic environments, is briefly reviewed. Measurements suggest that ocean wave-generated noise at the periphery of the ice-covered Arctic Ocean is generally exceptionally high when the fetch is large, and can have very long range effects, when propagated signals are not subject to significant surface scattering and volume absorption losses. Surface scattering due to sea-ice ridges, which may be modeled as randomly distributed elliptical half-cylinders, dominates under-ice propagation at frequencies above about 20 Hz. Ray theoretical calculations incorporating this model are consistent with observed site-dependent transmission loss differences over a limited frequency range (40–1000 Hz). Results of new lower frequency wave theoretical calculations, which incorporate this model and a suitable geoacoustic structure of

the ocean bottom, will be presented and compared with data. Initial concepts for extension of this work to three dimensions (including randomly oriented ridges) to enable more refined predictions of scattering/transmission losses and, in particular, spatial coherence losses will be outlined.

3:15

K5. Measurements of the vertical fields of acoustic signals and normal mode propagation in the Arctic Ocean. T. C. Yang (Code 5123, Naval Research Laboratory, Washington, DC 20375)

The vertical sound fields of cw signals and shots propagating in the ice-covered Arctic Ocean were measured during the FRAM IV experiments in April 1982. The measurements were made with a vertical array consisting of 28 hydrophones extending to a depth of 960 m. This experiment marked the first successful deployment of a large aperture vertical array in the Arctic environment. This paper will present the results and interpretations of the vertical array data. The lowest order normal mode amplitudes and phases were measured from the shot signals received simultaneously on the 28 hydrophones of the vertical array. The measured mode amplitudes are compared with the computed normal mode and used as the basis for spatial filtering of normal modes from the cw signals. Intensity distributions of the cw signals versus depth are presented for several frequencies below 100 Hz. Spatial coherence of the cw signals are studied and compared with that of the ambient noise. The arrival angles of the lowest order normal modes are measured from the cw signals using the mode filtering technique. The angular arrival pattern of the cw signals are plotted using conventional beamforming. Temporal behavior of the cw signals will be shown; phase tracking beamforming is used to enhance the vertical array gain.

Contributed Papers

3:40

K6. Seismic events in the Arctic Ocean. Ruth E. Keenan (Science Applications, Inc., Woods Hole, MA 02543) and Ira Dyer (Department of Ocean Engineering, MIT, Cambridge, MA 02139)

We present temporal and spectral characteristics of abyssally generated *T* phases observed in the Arctic Ocean. A large (~ 1 km) horizontal hydrophone array suspended through the ice made it possible to verify the event as seismic. A simple propagation model accounts for the basic features of each event. We identified crustal and water wave arrivals, and the time difference between the longitudinal crustal wave and peak frequency arrival of the *T* phase established the range to the source. The intersection of the bearing of the *T* phase with the mid-Arctic ridge provided a corroborating range estimate. *T*-phase sonograms display characteristic pear shapes noted by others. The spectra have peak levels at 5 and 15 Hz and behave asymptotically as f^4 , f^{-4} above and below their peak level. The duration of the *T* phase is longer than can be consistently accounted for by any one mechanism. Possible mechanisms including source depth, multiple bounces, dispersion, and swarms are discussed. [Supported by SAI, Inc. and ONR.]

3:55

K7. Icequakes in the Arctic Ocean. Peter J. Stein, Nicholas C. Makris, and Ira Dyer (Department of Ocean Engineering, MIT, Cambridge, MA 02139)

Resembling earthquakes, stress cracks in a floating ice sheet due to effects of winds, currents, and thermal gradients can excite various forms of elastic/acoustic wave propagation. The acoustic emission of several such icequakes were recorded during experiments on pack ice over the eastern Arctic abyssal plains. These events occurred in the near field of a 1-km aperture, 24-channel hydrophone array. The signals were bandpass filtered from 1 to 80 Hz. A recursive technique was used to locate these events and estimate the group speeds. Spectral characteristics are present-

ed for different types of events. Interchannel correlation was performed to obtain distortion characteristics of the propagation path. Using these results, possible paths of propagation from source to receiver are discussed. [Work supported by ONR.]

4:10

K8. Some ambient noise measurements in spring ice, Pt. Barrow, Alaska. W. C. Cummings (Oceanographic Consultants, 5948 Eton Court, San Diego, CA 92122) and D. V. Holliday (Tractor, Inc., 9150 Chesapeake Drive, San Diego, CA 92123)

Spring ice conditions off North Alaska are confounded by dynamic changes resulting from lead openings and closings, rapid changes in degrees and kinds of ice cover, rapidly changing wind conditions, and drifting floes. The measurements were taken in the spring of 1982 using a 9-nmi telemetering link. [Work supported by the North Slope Borough.]

4:25

K9. TRISTEN/FRAM IV Arctic ambient noise measurements. R. Nielsen, F. R. DiNapoli, M. Fecher, D. Potter (Naval Underwater Systems Center, New London, CT 06320), and P. L. Stocklin (Analysis and Technology, North Stonington, CT 06359)

Approximately 250 h of multichannel ice camp ambient noise data were recorded by NUSC during the TRISTEN/FRAM IV experiment in the Barents Abyssal Plain, Arctic Ocean during April 1982. Single hydrophone data were recorded from various elements of both the Massachusetts Institute of Technology/Woods Hole Oceanographic Institute horizontal array and the Naval Research Laboratory vertical array. This paper characterizes the average ambient noise level, and its variance as a function of frequency (to 2500 Hz), depth, and time. [This work is funded by ONR Code 425-AR, Program Manager, R. Obracta.]

Session L. Musical Acoustics II: Intonation and Scales

Donald E. Hall, Chairman

Department of Physics and Music, California State University, 6000 J Street, Sacramento, CA 95819

Chairman's Introduction—2:00

Invited Paper

2:05

L1. The tuning of Harry Partch's 43 tones-to-the-octave just intonation scale, and its musical endowments and consequences. Danlee Mitchell (San Diego State University, Music Department, San Diego, CA 92116)

Harry Partch (1901–1974) is the American composer noted for his bold and unique departure away from 20th century tuning and performance practices of our present day European influenced music monolith. During the 1920's Partch formulated the theoretical parameters of his tuning system, based on an interlocking harmonic and subharmonic grid extended to the 11th partial. This system features a "micro" tuned scale of 43 tones-to-the-octave, and just harmonic relationships in ratios of successive integers 4-5-6-7-9-11 that surpass the resources of 12-tone Equal Temperament. Not content with a mere theoretical victory, Partch then devoted his entire life to the building of sculpturally beautiful musical instruments, and the employment of these many instruments in his theatrically oriented musical works. Mr. Mitchell, Partch's close associate for the last 20 years of his life, will present an intimate view of Partch's world.

Contributed Papers

2:35

L2. Perception of musical interval tuning. Donald E. Hall (Departments of Physics and Music, California State University, 6000 J St., Sacramento, CA 95819)

Pairs of tones have been generated under computer control and presented to musically trained subjects through headphones. Musical relevance is increased by using filtered pulse waves rather than sines. The stimulus intervals are harmonic (both tones heard simultaneously) rather than melodic. These intervals are mistuned in varying amounts in the vicinity of each traditional musical interval (e.g., around the 3:2 frequency ratio for the perfect fifth). Judgments of the relative degree of perceived mistuning are recorded and analyzed; although related to judgments of "dissonance," these differ in significant ways. From the results we can infer relative ability to detect mistuning of different interval species. We find important differences from the traditional "consonance rank" ordering, and point out some implications for theories of keyboard tuning.

2:50

L3. Perception of melodic and harmonic intonation of two-part musical fragments. Rudolf A. Rasch (Institute of Musicology, Drift 21, 3512 BR Utrecht, The Netherlands)

Short musical fragments consisting of a melody part and a bass part were mistuned in various ways and to various degrees. They were presented to a group of subjects for a judgment of the quality of intonation and for an identification of the mistuned part. Mistuning was applied to the melodic frequency intervals of the melody part and/or the bass part (melodic mistuning) as well as to the frequency intervals of simultaneous tones in melody and bass (harmonic mistuning). Results indicate that the melodic mistuning of the melody part has the largest influence on the perception of the intonation, followed by the harmonic mistuning, while the melodic mistuning of the bass is less important. When mistuning is present in both melody and bass part, mostly the melody is considered to be the mistuned part. When the sound-pressure levels of the two parts are unequal, there is a tendency to consider the louder part as the mistuned part. In the perception of harmonic mistuning, the deviating interval size is probably of more

importance than the presence of beats. Small amounts of mistuning go unnoticed or may even cause an increase of the intonation quality.

3:05

L4. Phantom beats in mistuned chords. J. L. Hall, M. V. Mathews, and L. A. Roberts (Bell Laboratories, Murray Hill, NJ 07974)

Beats are heard in slightly mistuned musical intervals. For example, a mistuned fifth with frequencies $f_1 = 804$ Hz and $f_2 = 1204$ Hz will produce 4 beats per second. These beats have been attributed to interaction between the 3rd harmonic of f_1 at 2412 Hz and the 2nd harmonic of f_2 at 2408 Hz. However, 4 beats per second can be heard when the mistuned fifth is played with pure sinusoids which have no harmonics. A second hypothesis attributes these phantom beats to interaction between combination tones $f_2 - f_1$ at 400 Hz and $2f_1 - f_2$ at 404 Hz. Experimental results supporting the second hypothesis are presented. If one combination tone is removed by a cancellation tone of appropriate amplitude and phase, the beats can no longer be heard. If tones are added to a perfectly tuned fifth so as to approximate the combination tones produced by a mistuned fifth, the resulting stimulus is difficult to distinguish from the mistuned interval. Similar results were obtained for mistuned fourths, attributable to interaction between combination tones $f_2 - f_1$ and $3f_1 - 2f_2$.

3:20

L5. Modality and suffix effects in memory for music. Linda A. Roberts (Bell Laboratories, Murray Hill, NJ 07974 and Rutgers University, New Brunswick, NJ 08903), David R. Millen, Caroline Palmer (Rutgers University, New Brunswick, NJ 08903), and Vicky C. Tartter (Bell Laboratories, Murray Hill, NJ 07974 and Rutgers University, Camden, NJ 08102)

Three experiments were conducted to explore the well-documented modality and suffix effects found in language, using another dual-code symbol system: music. In all three experiments, subjects were tested for immediate serial recall of musical notes. In experiment 1, moderately and highly trained musicians recalled random orderings of notes contained within the span of an octave (D to D). The highly trained musicians dem-

onstrated a recency effect for both visual and auditory presentation. Moderately trained subjects demonstrated the effect typically observed for language: a recency effect was found only for the auditory condition. In experiment 2, the target notes were from a musically restricted range (D,F,A) and the rate of presentation was increased to discourage linguistic

recoding of stimuli. Results indicated a recency effect for both modalities and an overall auditory modality advantage. Results of experiment 3 (visual presentation only) showed a larger recency decrease (suffix effect) for an appended visual note than for a tone or a written letter. A dual-code representation of music is proposed.

TUESDAY AFTERNOON, 8 NOVEMBER 1983

CHAMBER ROOM, 1:30 TO 4:20 P.M.

Session M. Engineering Acoustics II: Numerical Techniques and Others

Roger L. Kerlin, Chairman

Applied Research Laboratory, Pennsylvania State University, P. O. Box 30, State College, Pennsylvania 16801

Chairman's Introduction—1:30

Contributed Papers

1:35

M1. The finite element method in acoustics. Earl Geddes (Electrical and Electronics Division, Ford Motor Company, Dearborn, MI 48121)

From the variational field equations for problems in acoustics one can derive the finite element approach to their solution. From this approach the application of a standard finite element method (FEM) program to a problem in room acoustics was performed. The role of the acoustic variables, the material constants, boundary conditions, etc., as required for the FEM program was examined. The need for complex arithmetic in the analysis of a nonconservative system and in the modeling of real sources was shown. The FEM eigenvectors and eigenvalues were then used to assemble the Green's function in the form of a data base. This Green's function was then to be used to examine the acoustic behavior of the room at low frequencies, i.e., below the Schroeder frequency.

1:50

M2. The addition of piezoelectric properties to structural finite element programs by matrix manipulations with applications to underwater acoustics. Graham McDearmon, Lyle Pauer, and Richard Scherch (Goodyear Aerospace Corporation, Department 456G2, 1210 Massillon Road, Akron, OH 44315)

The capability to analyze materials having linear piezoelectric properties was developed for structural finite element programs. Of the many methods with which this could be achieved, the method chosen was to add the piezoelectric properties by matrix manipulation of elastic and heat transfer element matrices. This method is independent of the elemental shape functions. It therefore has the advantages of converting many varieties of structural finite elements into piezoelectric finite elements with one general coding and as the state of the art improves for the structural elements so will the piezoelectric ones without additional or new programming. Otherwise, each variety of piezoelectric element and improvements would have to be programmed separately. This capability was added to MSC/NASTRAN by Goodyear Aerospace Corporation. The theory behind this method and comparison of finite element models of several piezoelectric ceramic underwater acoustic projectors to experimental data are presented.

2:05

M3. Analysis of intermodal coupling in piezoelectric ceramic rings. George W. Benthien (Code 7122, Naval Ocean Systems Center, San Diego, CA 92152) and Gordon E. Martin (Martin Acoustics Software Technology, P. O. Box 86050, San Diego, CA 92138-6050)

Many longitudinal vibrators employ a multi-element stack of piezoelectric ceramic rings as the active component. In order to design such transducers it is desirable to be able to measure longitudinal 33-parameters of piezoelectric ceramics using a single-ring geometry. Existing one-dimensional models are not adequate to describe the "longitudinal" mode for typical rings due to intermodal coupling. A piezoelectric finite element model was used to relate resonant frequencies to 33-ceramic parameters for certain selected ring geometries. However, since piezoelectric finite elements are not generally available, a simplified model was developed in which the necessary computations can be performed on a hand calculator. Guided by the finite element results, idealized mode shapes were used in conjunction with an energy method in order to obtain the simplified model. The resonant frequencies predicted by the simplified model were in excellent agreement with those obtained using the finite element method. This new model can be used to analyze a broad range of ring geometries. The authors have used the aforementioned models to obtain 33-piezoelectric parameters from measurements of resonant frequencies and low-frequency capacitance on ceramic rings.

2:20

M4. An improvement of the finite radiating element formulation—Application to the modeling of a radiating free-flooded transducer. Régis Bossut (Sintra Alcatel-Département D. S. M., Z. I. des Paluds, 13400 Aubagne, France) and Jean-Noël Decarpigny (ISEN, 3 rue François Baès, 59046 Lille Cedex, France)

At low frequencies, the limit of a free-flooded cylinder farfield is very far from the acoustic center. Thus, modeling this submerged structure by the finite element method requires a large number of fluid elements, when the radiation boundary condition is represented by an outgoing spherical (monopolar) wave. A radiating element has been created, that uses a radiating impedance of both monopolar and dipolar waves. With this new element, the radiating surface can be much closer to the transducer, and then the number of fluid elements decreases significantly. Unfortunately, one cannot obtain directly the farfield directivity pattern, because the radiating surface is inside the nearfield zone. However, it is possible to find it by a simple algorithm. This algorithm can also be used in an acoustical tank whenever its size is too small to measure the far acoustic field of a large transducer.

2:35

M5. Efficiency enhancement of thermo-acoustic sources through high-velocity motion. N. P. Chotiros (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78712)

A theoretical study of thermo-acoustic sources in the context of sonar applications was undertaken. The main objectives were to improve energy efficiency and to provide a means for beamsteering, using a moving thermo-acoustic source. The results show that improvements in energy efficiency of several orders of magnitude are possible at very large Mach numbers. (This is quite distinct from peak intensity efficiency which occurs at Mach 1.) The results also show the possibility of beamsteering by moving the thermo-acoustic source in a number of ways. The results suggest that the thermo-acoustic source may, in certain sonar applications, be a viable acoustic projector. [Work supported by Office of Naval Research.]

2:50

M6. An application of SEA to helicopter structure borne noise. D. L. Hawkins (Engineering Division, Westland Helicopters Ltd, Yeovil, Somerset, England)

The paper describes an exploratory application of SEA to helicopter structure borne cabin noise. The work was focused on a full-scale center cabin test section of advanced structural design, rather than a complete helicopter fuselage. Several theoretical models were developed, differing in the complexity of the structural representation adopted. The main difficulties encountered in the modeling process arose from the composite sandwich construction of the wall panels, and in particular expressing the coupling loss factors from the beamlike frame elements to these panels. Numerical predictions based on these theories were made of the expected vibration levels around the structure, and the acoustic pressure within the cabin. These predictions were compared with measured results from the trial structure. Good overall agreement was found with the more refined model, although the cruder model still gave acceptable prediction of noise levels across the frequency range considered (250–8000 Hz). It is concluded that SEA is potentially a valuable tool for predicting helicopter cabin noise. [Work supported by U. K. MOD.]

3:05

M7. Surface intensity and radiation loading on cylindrical surfaces via FFT methods. P. Stepanishen and H. W. Chen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

A numerical approach is presented to evaluate the surface intensity and radiation loading on a finite cylindrical surface with a known harmonic radial velocity distribution. The approach is based on a combined FFT and Green's function method [P. Stepanishen and H. W. Chen, *J. Acoust. Soc. Am. Suppl.* 1 73, S22 (1983)]. Surface pressure, intensity, and radiation loading are rapidly obtained via the use of standard FFT methods. The spatial fluctuations of the surface pressure and intensity for a known velocity distribution can thus be readily investigated. Numerical results are presented to illustrate the spatial variation of surface pressure and intensity as a function of axial mode shape, circumferential mode number, and frequency. Negative intensity regions and edge effects are clearly observed in the results at low frequencies.

3:20

M8. A temperature correlation for the radiation resistance of a thick-walled circular duct exhausting a hot gas. J. R. Mahan (Department of Mechanical and Aerospace Engineering, West Virginia University, Morgantown, WV 26506), J. G. Cline (General Dynamics Corporation, Fort Worth, TX 76101), and J. D. Jones (Mechanical Engineering Department, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

It is often useful to know the radiation impedance of an unflanged but thick-walled circular duct exhausting a hot gas into a relatively cold surroundings. Results based on data available in the literature [N. Fricker and C. A. Roberts, *Acustica* 38, 124–130 (1977); A. Cummings, *J. Sound Vib.* 52, 299–304 (1977)] and a new experimental study confirm that the reactive component is insensitive to temperature, while the resistive component increases with the temperature difference between the gas flowing from the duct and the surrounding air. A temperature correlation is developed permitting prediction of the radiation resistance from a knowledge of this temperature difference. The effect of wall thickness is shown to be

unimportant compared to other geometrical and acoustic factors for a range of thickness of practical interest. A physical basis is presented for the observed variation of the radiation resistance with temperature. [Work supported by NASA.]

3:35

M9. Transition from the nonlinear King integral to spherical propagation for a finite amplitude sound beam. Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The propagation of finite amplitude waves radiating from a baffled piston has been described in terms of directional spherical waves [J. C. Lockwood, T. G. Muir, and D. T. Blackstock, *J. Acoust. Soc. Am.* 53, 1148–1153 (1973)]. That analysis predicts waveforms at large distances, provided that comparable information is known at a reference location in the farfield. Lockwood *et al.* used this approach based on assuming that linear theory is accurate at the reference location. Such an assumption is inaccurate when the source pressure level is sufficient to generate significant nonlinear effects (growth of higher harmonics and depletion of the fundamental) within the near field. The present work describes the interfacing of the spherical propagation theory and the nonlinear King integral [J. H. Ginsberg, *J. Acoust. Soc. Am. Suppl.* 1 71, S30 (1982)]. The latter theory is used in this approach to evaluate the nonlinear waveform at the reference location. Comparing the results of interfacing, and of direct propagation according to the nonlinear integral formulation, with experimental data provides a strong validation for both theories. The advantage of using directive spherical wave theory lies in its superior computational efficiency and its ability to describe shock formation. [Work supported by ONR, Code 425-UA, and NSF, Grant MEA-8101106.]

3:50

M10. Sound reflection from a fluid-loaded and masked elastic plate. P. D. Jackins and G. C. Gaunaud (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20910)

We have extended available computer codes that predict the classical reflections and transmissions of sound fields through sets of plane elastic layers [i.e., D. Folds and C. Loggins, *J. Acoust. Soc. Am.* 62, 1102–1115 (1977)]. The present layers separate possibly dissimilar fluids, are possibly viscoelastic, and are isonified from one side. The results we display include reflection (R) and transmission (T) coefficients as functions of frequency f and incidence angles θ . This computational tool has been used to predict the reflections from a single fluid-loaded elastic plate ideally covered with a lossy layer of homogeneous viscoelastic material, isonified from the coated side. Resonances in the coefficients are analyzed in the light of the resonance scattering theory (RST), which has not yet been applied to coated plates. The present model serves to assess the effectiveness of the masking layer to reduce the reflections from the metal plate. The predicted displays of calculations are presented for values of the material parameters in the ranges commonly found in practice. The lossy nature of the absorbing layer complicates the otherwise real parameters present in the analysis, and we quantitatively determine the shifts in resonance locations, the widening of the resonance peaks, and the overall reduction of the returned reflections caused by the masker's viscoelasticity.

4:05

M11. Distribution of resonance frequencies of coupled dynamic systems. G. Maidanik and L. J. Maga (David W. Taylor Naval Ship Research and Development Center, Bethesda, MD 20084)

Finite, uncoupled dynamic systems have response maxima at specific frequencies, the resonance frequencies, the distribution of which can be ascertained. When the dynamic systems are coupled and, therefore, interact with each other, the coupling affects the distribution of the resonance frequencies. In this paper the dynamic systems are modeled as one-dimensional wave-bearing systems and the coupling is effected via the transmission coefficients of the junctions between the systems. The nature of the change in the distribution of the resonance frequencies is examined in some simple cases.

Session N. Architectural Acoustics III, Noise II, and Shock and Vibration II: Active Sound and Vibration Control

Oliver L. Angevine, Chairman

Angevine Acoustical Consultants, Inc., 7349 Davis Road, West Falls, New York 14170

Chairman's Introduction—1:00

Invited Papers

1:05

N1. Waveform synthesis—The Essex solution to cancelling periodic noise and vibration. G. B. B. Chaplin (Wolfson Centre for the Electronic Cancellation of Noise and Vibration, University of Essex, Wivenhoe Park, Colchester CO4 3SQ, England)

The paper will give a basic description of the special method developed by the Essex Group for cancelling noise and vibration of a periodic nature. One of the advantages of the method is that the anti-noise or anti-vibration is produced by synthesis and does not depend on measurement of the original noise or vibration. The anti-noise and anti-vibration can therefore be injected at precisely the right time since no signal processing delay is involved. Examples will be given of applying the system to cancelling the noise at the exhaust of a diesel engine and the vibration at its mounting points, and to cancelling noise within a cab. The system is also highly selective and it is possible to cancel one source of noise whilst leaving other noise sources untouched. Experience on cancelling the exhaust stack noise on a cargo ship will be discussed.

1:35

N2. Active noise cancellation in ducts. Jiri Tichy (The Pennsylvania State University, State College, PA 16802), Glenn E. Warnaka (Lord Corporation, 2000 West Grandview Boulevard, P. O. Box 10038, Erie, PA 16514-0038), and Lynn A. Poole (The Pennsylvania State University, State College, PA 16802)

This paper summarizes the state of active noise cancellation in ducts. The application of active feedback to reduce the sound energy propagation through the duct is primarily important at low frequencies because passive silencers are very large, and the attenuations achieved are relatively small. In its classical configuration, the noise field detected by a signal microphone is phase inverted and time delayed by an adaptive filter. This signal is then reradiated into the duct from a sound source at the duct wall or located remotely and connected by means of a waveguide. The adaptive filter is controlled by a microphone located in the duct to sense the noise reduction and adjust the entire system for optimum operation. This paper analyzes the effects of the signal microphone position, type, and location of the cancellation sound sources, and sensing microphone location. Optimum operation requires consideration of higher order modes and the nearfield of the cancellation source, including evanescent waves. Results obtained with practical constructions illustrate the great potential of active systems to attenuate noise with discrete spectra, random noise, as well as transients are presented.

2:05

N3. Active control of noise in enclosed spaces. Glenn E. Warnaka, John M. Zalas (Lord Corporation, 2000 West Grandview Blvd., P. O. Box 10038, Erie, PA 16514-0038), Jiri Tichy, and Lynn A. Poole (The Pennsylvania State University, State College, PA 16802)

Active methods for controlling noise have been demonstrated to be feasible. As they are compact in volume and low in weight, active controls seem very promising for controlling low-frequency noise in enclosed spaces, particularly in transportation vehicles. Nevertheless, certain difficulties arise due to the modal structure developed within the enclosed spaces, because the cancellation sources must produce an exact match of the unwanted noise field in order to cancel it. Similarly, the nonuniform nature of the noise field poses problems in detection of the noise. This paper discusses the progress which has already been made in controlling interior noise and shows that there are several approaches that may be adopted. The methods depend on whether the noise originates within the space or from sources outside the space. If the sound enters the space by penetration from the outside, then the method of penetration can be important in determining the location of the cancelling sources and the detecting microphones. The technical potential and the latitude of application indicate that there is considerable promise for active control of noise in enclosed spaces.

2:35

N4. Active systems for global attenuation of noise. M. J. M. Jessel (CNRS-LMA, B. P. 71, F-13277 Marseille Cedex 9, France) and O. L. Angevine (Angevine Acoustical Consultants, Inc., 7349 Davis Road, West Falls, NY 14170)

Theoretical approaches to active noise absorption [G. A. Mangiante, *J. Acoust. Soc. Am.* **61**, 1516–1523 (1977)] provide criteria that may prove useful for the design of any system for active noise attenuation, especial-

ly one which must produce attenuation in all directions around a complex sound source. (1) *Topological fitness*: a proper net of "anti-sources" ought to surround completely either the primary noise source(s), or the space to be silenced. (2) *Directivity*: each anti-source ought to radiate only towards the space to be silenced. (3) *Concentration*: practical system design requires substituting the theoretical continuous distributions of anti-sources by discrete ones. This approach has been successfully applied to reducing the audible hum of an actual electric substation transformer. The active abatement method is easier for discrete frequencies, such as the spectrum of transformer hum, and is most effective at the relatively low frequencies—120, 240, 360, and 480 Hz—most prominent in the transformer hum spectrum. A relation has been found experimentally for the attenuation achievable versus the number of sources for each of these hum frequencies. The active attenuator interferes less with the normal ventilation and cooling of a transformer than do passive methods. Active abatement also allows attenuation differing in different directions.

3:05

N5. Coherent active methods for applications in room acoustics. Dieter Guicking (Drittes Physikalisches Institut der Universität, Buergerstr. 42-44, D-3400 Goettingen, West Germany)

An adjustment of reverberation time in rooms is often desired even for low frequencies where passive absorbers fail. Among the active (electroacoustic) systems, incoherent ones such as "assisted resonance" permit prolongation of reverberation time only, whereas coherent active methods—as proposed by H. F. Olson since 1953—allow sound absorption as well. A coherent-active wall lining consists of loudspeakers with microphones in front and adjustable control electronics. The microphones pick up the incident sound and drive the speakers such that the reflection coefficient takes prescribed values. An experimental device for the one-dimensional case has been developed and allows reflection coefficients between almost zero and about 1.5 to be realized below 1000 Hz, employing moving coil or specially developed extremely flat electret loudspeakers. The extension to three dimensions is being investigated; problems arise mostly from diffraction effects. The actual state of the experimental work including a comparison with model computations will be outlined. Furthermore, some ideas about active diffusers are presented. In contrast to passive "Schroeder diffusers" with inherent frequency dependence, active diffusers are basically broadband systems.

TUESDAY AFTERNOON, 8 NOVEMBER 1983

FORUM ROOM, 1:30 to 4:50 P.M.

Session O. Physical Acoustics II: General Topics I

Steven L. Garrett, Chairman

Physics Department, Naval Postgraduate School, Monterey, California 93940

Chairman's Introduction—1:30

Contributed Papers

1:35

O1. Reciprocity calibration of "second sound" transducers in superfluid helium S. L. Garrett (Physics Department, Code 61Gx, Naval Postgraduate School, Monterey, CA 93940)

The reciprocity theorem has been used for over 40 years to facilitate absolute calibrations of electroacoustic transducers from purely electrical measurements, without reference to a primary acoustical standard. A generalization of this calibration procedure to reversible second sound transducers in He II will be described. The symmetry of two-fluid hydrodynamics is used to derive an expression for the second sound specific acoustic transfer admittance, thus permitting absolute mechanical measurement of temperature oscillations in superfluid helium. In the temperature range from 1.15° to 1.9°K, the Tisza approximation and the thermodynamics of an ideal Bose-Einstein gas [F. London, *Superfluids*, Vol. II, Sec. 7] lead to a transfer admittance which is quadratic in temperature. This generates a calibration constant that has the experimentally convenient feature of being a linear function of absolute temperature [$f(T) = 21.31T - 17.01 \pm 0.17 \text{ g}^{1/2} \text{ K}^{-1} \text{ s}^{-3/2}$]. Applications to non-linear hydrodynamics, photoacoustic spectroscopy, and the characterization of porous diaphragm second sound transducers will be suggested. [Work supported by the NPS Foundation Research Program.]

1:50

O2. An experimental test of acoustical reciprocity calibration applied to thermal transducers in superfluid helium J. Valdivia, Jr., B. R. Ogg, R. K. Yarber, and S. L. Garrett (Physics Department, Code 61Gx, Naval Postgraduate School, Monterey, CA 93940)

An experiment will be described which has verified an extension of the reciprocity calibration technique to reversible thermal transducers in superfluid helium. A plane-wave resonator of circular cross section was capped at both ends by reversible teflon electret diaphragms which contained many razor slits to generate or detect thermal waves (second sound). The resonator also incorporated a thermophone and a dc biased carbon resistance thermometer to set independent upper and lower limits on the amplitude of the temperature oscillations within the resonator. The acoustical resonance data was acquired and analyzed by a computer controlled system described previously [D. V. Conte and S. L. Garrett, *J. Acoust. Soc. Am. Suppl.* 1 72, S82 (1982)]. The temperature excursions measured by the reciprocity method fell between the upper and lower limits which, for lower modes, were separated by only a few percent. At higher modes the lower limit departed from the upper limit due to the thermal inertia of the resistance thermometer. Temperature excursions smaller than $10^{-8} \text{ K}/[(\text{Hz})^{1/2}]$ were detectable using the electret trans-

2:05

O3. Comparison of procedures for determination of acoustic nonlinearity of some inhomogeneous materials. L. Bjørnø (Industrial Acoustics Research Group, Technical University of Denmark, Building 352, DK-2800 Lyngby, Denmark)

The widespread use of ultrasonic energy in medical diagnosis and therapy and the recent research showing the existence of nonlinear acoustic effects in inhomogeneous materials, like biological media, have led to a strongly increased interest in investigations aiming at providing enough information for an evaluation of the second order nonlinearity ratio B/A as a tissue characterizing parameter on a par with attenuation, impedance, etc. The several techniques used for the determination of B/A roughly fall into two main groups, finite-amplitude methods and thermodynamic-acoustic methods. The B/A values determined using various techniques of the two main groups are strongly scattered, showing a dependence on the procedure used, which makes it difficult to draw any valid conclusions. The paper discusses advantages and disadvantages of the techniques used hitherto, aiming at an explanation of the reasons for the deviations between the present B/A results for biological media. The necessary accuracy in the determination and the reliability in the exploitation of B/A as a tissue characterizing parameter for clinical use are emphasized.

2:20

O4. Ultrasonic evidence for structural relaxation in large unilamellar liposomes. Philip R. Strom-Jensen, Richard L. Magin, and Floyd Dunn (Bioacoustics Research Laboratory, University of Illinois, Urbana, IL 61801)

The ultrasonic absorption of large unilamellar vesicles (average diameter $0.2 \mu\text{m}$) was determined in the frequency range 0.5–5 MHz. The liposomes were composed of a 4:1 mixture by weight of dipalmitoyl phosphatidylcholine and dipalmitoyl phosphatidylglycerol. A large increase in absorption occurs at the solid to liquid-crystalline phase transition temperature (42°C) of the pure lipid vesicles. This increase in absorption is interpreted as a structural relaxation of the "melting" fatty acid chains occurring with an average relaxation time of 76 ns. The liposomes were also found to be extremely permeable near the transition temperature, releasing encapsulated cytosine arabinoside. Addition of cholesterol or gramicidin to the liposome bilayer broadened the ultrasonic absorption and reduced the efflux of cytosine arabinoside at the transition. Gramicidin, in addition to broadening the transition, slows the isomerization of bonds in the hydrocarbon chains of the lipids. A concentration of 5 mole percent gramicidin increased the average relaxation time to 210 ns. [Work supported in part by grants from the National Cancer Institute, National Institutes of Health, and the American Cancer Society.]

2:35

O5. Nearfield of a piston source in an absorbing medium. Wesley L. Nyborg and R. Bruce Steele (Department of Physics, University of Vermont, Burlington, VT 05405)

Medical exposures to ultrasound from unfocused source transducers usually occur at distances from the source which are not great enough to justify a farfield approximation. In order to analyze biophysical interactions which occur in the exposed medium, it is necessary to consider details of the nearfield, taking absorption into account. A formulation, in terms of a line integral, given by Schoch [A. Schoch, *Akust. Z.* **6**, 318–326 (1941)] for a piston source proves to be useful for this purpose. A paraxial approximation which he derived leads to an analytical expression which applies even to a central region of the important, but complex, field which exists just in front of the source. In this paper computed plots will be shown, comparing results obtained by use of solutions and approximations given by Rayleigh, Schoch, and others. These comparisons give information on the conditions under which various approximations are valid, and on the influence of absorption on the field distributions. [Work supported by the National Institutes of Health, Research Grant GM-08209.]

O6. Ultrasonic study of oil/water emulsions stabilized with nonionic block copolymers. M. A. Barrett Gultepe, M. E. Gultepe, J. L. McCarthy, and E. B. Yeager (Ultrasonic Research Laboratory, Department of Chemistry, Case Western Reserve University, Cleveland OH 44106)

An increment in sound velocity at 2 MHz has been observed as the emulsion globule size decreases with time in a water, toluene, and polyoxyethylene (POE)—polyoxypropylene (POP) block copolymer system. The observed effect depends on the copolymer concentration and is largest for the block copolymer composed of 50% POE–50% POP moieties. A general acoustical treatment taking into account viscous loss in the emulsions globules as well as surrounding liquid and heat conduction across the interface fails to explain the observed velocity increment. Aqueous solutions of the above copolymer shows association effects as evidenced by a break in the sound velocity versus concentration curve at approximately 15 g/l. [Work supported by ONR.]

3:05

O7. Sound absorption in aqueous mixtures of MgSO_4 and HCl. C. C. Hsu (Department of Electrical Engineering, Chung Cheng Institute of Technology, Taoyuan, Taiwan 335, Republic of China) and F. H. Fisher (Marine Physical Laboratory of the Scripps Institution of Oceanography, University of California at San Diego, San Diego, CA 92152)

Sound absorption measurements were made at 25°C over the frequency range of 30–300 kHz for aqueous mixture of 0.02 M MgSO_4 and HCl, using a spherical resonator method. At 1 atm, the excess sound absorption measured for a mixture of 0.1 M HCl is negligible while that of 0.05 M HCl showed a maximum absorption, $(\alpha\lambda)_{\text{max}} \times 10^6$, of 9.49 ± 0.21 at 214 kHz and that of 0.01 M HCl, 68.90 ± 0.71 at 161 kHz. At 307 atm the mixture of 0.05 M HCl showed a maximum absorption of 5.78 ± 0.14 at 205 kHz. These results will be discussed in terms of normal mode coupling of HSO_4^- and MgSO_4 dissociation reactions. The observed decrease in absorption as HCl is added cannot be explained by ion-pairing concepts alone.

3:20

O8. Shape oscillations of microparticles on an optical microscope stage. Zhu Zhe-ming¹ and Robert E. Apfel (Yale University, P. O. Box 2159, New Haven, CT 06520)

Forces arising from modulated acoustic radiation pressure have been used in the past to produce shape oscillations of drops in order to study surface properties [e.g., P. L. Marston and R. E. Apfel, *J. Acoust. Soc. Am.* **67**, 27 (1980) and C. J. Hsu and R. E. Apfel, *J. Acoust. Soc. Am.* **70**, S90 (1981)] and to study the oscillating drop's internal flow patterns [E. Trinh, A. Zvern, and T. G. Wang, *J. Fluid Mech.* **115**, 453 (1982)]. We have extended this work to study surface properties of much smaller particles (8 to 200 μm , approximate diameter) by fabricating a stage for an optical microscope that incorporates a 10-MHz, piezoelectric, quartz disk electrode in such a way as to allow the transmission of light through a central region. The quartz disk forms part of a thin chamber into which the microparticles in a host liquid can be injected. Establishing an acoustic standing wave in the chamber has enabled us to study shape oscillations for drops resting on a surface of the chamber and for drops acoustically levitated away from solid container surfaces. At high acoustic amplitudes drop fission has been produced. We shall present data and a photographic record for hydrocarbon drops in water. We hope also to present results for biological cells, since the investigation of the properties of their membranes was the original motivation for this work. [Work supported by the National Institutes of Health, Grant 1R01-GM30419, and the U. S. Office of Naval Research.]¹ Also at Nanjing University, People's Republic of China.

3:35

O9. Sample transport of levitated objects in a dual temperature resonance chamber. J. Robey and E. Trinh (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

Experimental studies show the feasibility of transporting an acoustically levitated object in an elongated resonance chamber filled with a gaseous medium, and having its opposite end zones at widely different temperatures (35° and 500 °C). Theoretical considerations describe mode behavior and the conditions for optimum power transfer into the heated region as well as the dynamic pressure distribution for various simple and complex modes. By properly introducing selected combinations of these modes, positioning and movement along the longitudinal axis and through the temperature gradient are possible. The reduced gravity of space would provide a useful environment for application of such a system in the areas of material processing, although the experimental work here has been carried out in the laboratory under 1 g with low-density materials.

3:50

O10. Method for separation of acoustic levitation and rotation modes leading to arbitrary axes of rotation J. L. Allen and M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

A method for separating acoustic levitation and rotation of small objects has been developed and tested experimentally. Sample rotation in an acoustical resonance chamber requires a phase difference between two equal frequency (degenerate) modes. Therefore, this method uses one set of nondegenerate modes for levitation and higher order sets of degenerate modes for rotation. This separation of modes may be accomplished by appropriate choices of chamber dimension ratios producing up to three sets of degenerate orthogonal higher order modes. By appropriate excitation of these sets of modes, arbitrary axes of rotation may be achieved. This technique for one rotation axis was verified experimentally using a rectangular chamber. Spherical samples were stably levitated without rotation using the three orthogonal fundamental modes, and rotated about the z axis by generating a phase difference between the degenerate third and fifth harmonics of the x and y directions, respectively. A video tape of these levitation experiments will be presented. [Work supported by NASA.]

4:05

O11. Acoustic force potential on a sphere in rectangular, cylindrical, and spherical resonators. M. Barmatz (Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109) and P. Collas (Department of Physics and Astronomy, California State University-Northridge, Northridge, CA 91130)

Generalized expressions of the acoustic force potential on a sphere in rectangular, cylindrical, and spherical geometry have been developed. The method of Gor'kov [L. P. Gor'kov, Sov. Phys. Dokl. 6, 773 (1962)], valid for arbitrary sound fields was used to derive these expressions in the limit of small sample radius ($kr \ll 1$). The critical points of the force potential, where all the force components are zero, were investigated. General expressions for the coordinates of the potential minima for various modes were determined from theoretical considerations and an analysis of computer generated potential contour graphs. Besides isolated minima points, nonisolated minima were found consisting of lines, circles,

and plane, cylindrical and spherical surfaces. The behavior of the acoustic pressure, potential, and forces for selected modes from each geometry is presented. The application of these modes to the field of acoustic levitation is also discussed. [Work supported by NASA.]

4:20

O12. Propagation of standing waves in axisymmetric cavities enclosing an inhomogeneous medium. M. El-Raheb and P. Wagner (Applied Mechanics, Jet Propulsion Laboratory, California Institute of Technology, Pasadena, CA 91109)

This paper studies how gradients in speed of sound c and density ρ influence standing waves in an axisymmetric cavity. The c and ρ gradients are assumed independent of radial and circumferential coordinates of the cavity and to vary only along its axis. The effects on propagation of the c and ρ gradients are studied separately. First, the problem with a step function as c gradient in a cylindrical cavity is treated in detail since it is amenable to analytical solution. A smooth c gradient described by exponential functions is considered. In this case, a transfer matrix coupled to a numerical integration procedure is adapted to the solution of the differential equation governing the axial dependence. The quasilinear c^2 gradient follows and it also yields to analysis. The poor energy transmission characteristics of transverse standing waves in cylindrical cavities with c gradient prompted consideration of a geometry that includes two cylindrical segments with different cross sections joined by a conical connector. The transfer matrix method couples the acoustics of various segments of cavities. The radial dependence in the conical segment is determined by the method of shooting while the axial dependence follows from numerical integration. The analysis is then extended to cavities with concentric center bodies. An approximation considers piecewise conical frustra and leads to discontinuous pressure gradients at the junction of conjoined segments. The approximate analysis is tested against numerical results from an accurate simulation that invokes a Green's function and surface elements with account made for medium inhomogeneity.

4:35

O13. Test of the nearfield holography technique using an un baffled, uniformly oscillating disk. Todd B. Beyer and Julian D. Maynard (Department of Physics, Pennsylvania State University, University Park, PA 16802)

Experimental results of nearfield holography measurements of the radiation from an un baffled, uniformly oscillating disk have been compared with the exact theoretical solution. The experimental source is a disk 1 m in diameter constructed of an aluminum honeycomb material that is lightweight and structurally rigid. Eight drivers located beneath the disk can be individually adjusted so that the disk's surface vibrates in-phase with constant amplitude. The radiation efficiency determined from the measurements at two frequencies (150 Hz, 250 Hz) agrees closely with the radiation efficiency calculated from the exact theory. The theoretical solution makes use of oblate spheroidal functions to compute the pressure on the surface of an ideal rigid disk vibrating with uniform normal velocity. [Work supported by the National Aeronautics and Space Administration.]

Session P. Musical Acoustics III: Wind Instruments

R. Dean Ayers, Chairman

Department of Physics–Astronomy, California State University, Long Beach, California 90840

Chairman's Introduction—3:35

Contributed Papers

3:40

P1. The effects of the flaring bell on the normal frequencies of a trombone. J. Duane Dudley and William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

As sinusoidal waves encounter the bell of a brass wind instrument, they are effectively reflected by the flaring part of the bore. A given frequency is effectively reflected from that part of the bell where a significant change in bore radius occurs within a wavelength. Hence the longer wavelengths reflect from points closer to the inlet end of the bell than do the shorter wavelengths. A computer calculation of the equivalent lengths corresponding to the normal frequencies shows this effect, and shows how the frequency shifts depend on the shape of the bore. Through this effect, the bell makes an important contribution towards making the normal frequencies harmonic.

3:55

P2. Effect of the player's vocal tract on the tone quality of reed woodwind instruments. John Backus (Physics Department, University of Southern California, Los Angeles, CA 90089-1341)

It has been argued by woodwind instrument players that the configuration of the player's vocal tract has an effect on the tone quality of the instrument when played. This matter is being investigated. The following arrangement has been set up: A "clarinet" is simulated by mounting a clarinet mouthpiece and reed on a piece of brass tubing of such length as to sound the note D3 when blown, using a foam neoprene "lip" with the reed. A microphone is attached to the mouthpiece to monitor its internal pressure. The open end of the brass tube is sealed onto the end of a 4-in.-diam tube 10 ft long filled with sound absorbing material. By attaching a vacuum cleaner to the other end of the 4-in. tube, the "clarinet" can be "blown," as it were, in reverse, with the reed end out in the open. Various systems can now be attached to the mouthpiece and their effect on the internal waveform of the mouthpiece determined. Results so far indicate that the effect of the player's vocal tract on the instrument's tone quality should be negligible. [Work supported by the NSF.]

4:10

P3. Edgetone spectra. A. W. Nolle (Department of Physics, The University of Texas at Austin, Austin, TX 78712)

It is well known that as the velocity of an air jet impinging on an edge is increased, various regimes of periodic oscillation are found in which the frequency increases with jet velocity. It will be shown that also there are narrow ranges in which the signal consists of two or more components, not harmonically related, and various intermodulation products. Ultimately, as the jet velocity increases, the signal becomes a noise band. Extensive data on these phenomena will be shown. In the present work the jet-

to-edge distance is 4–12 mm, and the jet velocity lies between 5 and 30 m/s. In resemblance to a common practice in organ pipe construction, the piece whose edge is struck by the jet is unsymmetrical, consisting of a 45° wedge mounted so that one side is parallel to the center line of the jet.

4:25

P4. The conical bore in musical acoustics. R. Dean Ayers, Lowell J. Eliason, and Daniel Mahgerefteh (Department of Physics–Astronomy, California State University, Long Beach, CA 90840)

A review of textbooks and research literature reveals that there is room for improvement in the treatment of this simple but important bore shape. We use plots of acoustic pressure standing waves to show students in a descriptive course that a complete cone and an open pipe of the same length have the same natural frequencies. These plots also predict the qualitative behavior for the frequencies of a frustum closed at the small end and suggest a rigorous approach to their calculation. Incorporating the open end correction for a straight pipe plus a correction for viscosity and thermal conductivity in the calculation results in agreement with experimental values to within 0.5%. In order to understand the inharmonicity of the frustum's frequencies from a different approach, we have calculated the pressure impulse response for reflection from its closed end. We find that this consists of a delta function plus an inverted, exponentially decaying wake. Experimental observations confirm both that shape for the impulse response and its contribution to the evolution of the waveform through several reflections.

4:40

P5. Recorder multiphonics. R. Dean Ayers, Lowell J. Eliason, and David Forel (Department of Physics–Astronomy, California State University, Long Beach, CA 90840)

A systematic study has been undertaken to identify and analyze all possible multiphonics on a popular brand of plastic recorder (Aulos, alto voice). The absence of a bell at the end of the recorder combines with the open-hole cutoff effect to guarantee strong, nonharmonic bore resonances, and many of the forked fingerings produce stable multiphonics when blown somewhat hard. The more stable and aurally interesting multiphonics have two strong spectral components, one supported by the fundamental bore resonance and the other by the fourth or fifth resonance. Heterodyne descendants of these components are also found, but rarely with comparable strength. The less interesting multiphonics have just one strong component, often at the third resonance, with noticeably weaker components beating against it. Some fingerings will produce more than one multiphonic, depending on the breath pressure and/or attack used. One factor that may be significant for stability is the fact that the edge tone itself goes multiphonic at the blowing pressures used, without any feedback from the bore.

Session Q. Speech Communication II: Speech Recognition and Processing Systems

Hisashi Wakita, Chairman

Speech Technology Laboratory, Division of MELTEK Corporation, 3888 State Street, Santa Barbara, California 93105

Chairman's Introduction—2:30

Contributed Papers

2:35

Q1. A reference speech recognition algorithm: An approach to characterizing speech data bases. James L. Hieronymus and David S. Pallett (National Bureau of Standards, Washington, DC 20234)

One way to characterize the relative recognition difficulty of speech data bases is to use a reference speech recognition algorithm. This approach was first suggested by R. K. Moore at the Royal Signals and Radar Establishment. A preliminary algorithm for this purpose has been published [G. F. Chollet and C. Gagnoulet, Proc. 1982 IEEE ICASSP, 2026–2029 (1982)]. Currently, an isolated utterance reference algorithm under development in our laboratory includes automatic endpoint detection and a training technique which averages several tokens of each utterance. Mel scale cepstral coefficients are used in conjunction with dynamic time alignment in both training and recognition. Recognition accuracy and performance statistics are presented along with distance measures for best and second-best match. Histograms of the ratio of second-best to first-best scores are plotted as a measure of data base confusability. If the ratio histogram has considerable population near unity, there is significant danger of confusion. Other descriptors characterizing properties of the speech data base are presented. Preliminary results of research on confusability measures are reported.

2:47

Q2. One-pass, speaker-independent, isolated digit recognition system. George Vysotsky (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

The system described in this paper is an outgrowth of one presented earlier [J. Acoust. Soc. Am. Suppl. 1 72, S30 (1982)]. It is a combination of deterministic and probabilistic approaches to isolated word recognition. The most distinguishing features of the new system are a real-time, one-pass algorithm and the use of telephone-quality speech rather than broadband microphone speech. This new version is targeted to a single chip implementation with the requirement that the system uses no more than 256 bytes of RAM and 8000 bytes of ROM. The system makes decisions based on a single 20 ms frame of ten features (with no long term storage of previous frames). Preliminary segmentation and matching are combined to significantly reduce both the number of confusable words at each recognition point as well as the memory requirement for references obtained during the multispeaker training. The system was evaluated on a ten digit vocabulary (1000 tokens, 21 talkers) and showed an accuracy of 94.4% with a 3.1% error rate, and a 2.5% rejection rate. The effects of using a one-pass algorithm on total performance are also discussed.

2:59

Q3. Speaker-independent isolated word recognition for small-size hardware using multiple valued Walsh Mel-cepstrum. M. Watari (C&C Systems Research Laboratories, NEC Corporation, 4-1-1 Miyazaki, Miyamae-Ku, Kawasaki-City, Kanagawa 213, Japan)

In this paper, the multiple-valued Walsh Mel-cepstrum is proposed as a speech parameter for a small-size speech recognizer. The new speech analysis is based upon the multiple-valued Walsh transform instead of the Fourier transform used in the standard Mel-cepstrum. This makes it possible to obtain good speech parameters by using a small microcomputer.

The analysis process has four stages: the multiple valued Walsh transform, the log magnitude computation, the Mel-scale conversion, and the inverse multiple valued Walsh transform. The multiple-valued Walsh transform operator uses multiple-valued coefficients ($p/2^n + q/2^j$) corresponding to $e^{j\omega}$ used in the Fourier transform. This operation requires only additions, subtractions, and shifts. Speaker-independent isolated digit recognition experiments were carried out to evaluate the new speech analysis method. The recognition error rates for Walsh Mel-cepstrum, multiple-valued Walsh Mel-cepstrum (16 valued), and Mel-cepstrum were 6.4%, 3.2%, and 3.1%, respectively.

3:11

Q4. A syllable-based connected-digit recognizer for telephone speech. D. Kahn (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

This paper describes a system for the automatic recognition of digits spoken in connected strings. The input is standard telephone-quality audio, bandlimited to 200–3200 Hz, spoken in a room with a background noise level of 60 dBA. The first step performed is a segmentation of the incoming string into syllables. Important features of the syllabification algorithm [D. Kahn, J. Acoust. Soc. Am. Suppl. 1 73, S88 (1983)] are (a) the analysis is based solely on the energy contour, with no spectral information, and (b) syllables judged to be unstressed are appended to the preceding stressed syllable; this is relevant only in the case of "seven," and results in the syllabic analysis being equivalent to a segmentation into digits. The remaining analysis centers around an Itakura-type dynamic-programming match between each extracted segment and a set of stored digit templates. The templates are themselves derived from segments extracted from telephone-quality connected-digit strings.

3:23

Q5. Evaluation of a signal-averaging method for the measurement of signal-to-noise ratios in speech. James Hillenbrand (Department of Communicative Disorders, Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

Many abnormal voice qualities are characterized by unusually large amounts of noise. Although the presence of noise is easy to identify, precise quantification of signal-to-noise (S/N) ratios is a difficult problem. Yumoto, Gould and Baer [J. Acoust. Soc. Am. 71, 1544–1550 (1982)] developed a technique based on signal averaging. Using a sustained vowel segmented into pitch periods, the signal component of the S/N calculation is defined as the average of the individual pitch periods. The noise component is estimated by successive subtractions of the average signal from individual pitch periods of the original vowel. In the present study, the accuracy of this technique was evaluated in a series of computer simulations. A synthesized vowel was mixed with varying amounts of synthesized noise. S/N ratios varied over a 36-dB range. Known S/N ratios were compared with S/N ratios calculated by the Yumoto *et al.* technique. The correlation between calculated and actual S/N ratios was nearly perfect ($r = 0.99$) and the mean absolute error was only 0.1 dB. Results are also reported for tests involving naturally produced vowels mixed with synthesized noise. [Work supported by NIH.]

Q6. The SIGPRO signal processing software development system. Jan I. Wolitzky (Bell Telephone Laboratories, Inc., Murray Hill, NJ 07974)

The SIGPRO system is designed to assist in the development of digital signal processing software. By freeing the user from such "bookkeeping" tasks as file management, memory allocation, control flow decisions, variable name and default parameter assignment, and error- and type-checking and correction, the SIGPRO system allows a signal processing software system to be specified solely in terms of standard "building block" modules and the connections between them. A library of approximately 100 such modules, for performing such operations as correlation, filtering and filter design, Fourier transforms, LPC synthesis, etc., is included, along with a mechanism for easily adding new modules to the system. The system is written entirely in the C programming language, and generates as its output a C program that can be compiled to produce correct implementation of the specified system, either as an executable program or as a C-callable subroutine (that may, in turn, be integrated into the module library). The program runs under virtual-memory versions of the Unix(TM) operating system. (Unix is a trademark of Bell Telephone Laboratories, Inc.)

3:47

Q7. Storage and retrieval of quality digitized long speech phrases using a microcomputer. Henry S. Todd (Department of Computer Science, Brigham Young University, Provo, UT 84602)

Two microcomputer digital speech storage and retrieval systems were designed and evaluated in a computer-assisted instruction environment.

An 8-bit Z80 microprocessor was used for logic control with storage of programs, text files and digitized speech on hard disk. Speech sampling rates of 10 and 12.5 kHz were utilized for digital 8-bit (256 amplitude levels) speech storage and reconstruction. Speech phrase lengths varied from 2 to 20 s.

3:59

Q8. A portable phonetics laboratory. Hector Javkin and Robert van der Veen (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

PIPS (Portable Instrumental Phonetics Station) is a portable box designed for gathering and analyzing aerodynamic and other phonetic data in the field. It contains two pressure transducers mounted 10 cm apart in a single tube with accompanying amplifiers, a high-speed chart recorder, a Rothenberg airflow measurement system, and a miniature storage oscilloscope. Circuits constructed at UCLA provide (1) preamplification to strengthen weak signals, (2) amplification to drive the chart recorder galvanometers (3) rms energy envelopes of ac signals, and (4) FM modulation and demodulation that can record and play back one channel of relatively high-frequency data or two channels of low-frequency data on a single track of a portable tape recorder. An experimenter normally uses the other track of a stereo tape recorder for an audio recording, although the two types of FM encoders can be used simultaneously to provide up to three channels of FM recording on a stereo tape recorder. Rechargeable batteries can drive most of the circuits. At present, the PIPS has been used in the U.S., China, and the Southern Sudan. [Work supported by NIH and NSF.]

TUESDAY AFTERNOON, 8 NOVEMBER 1983

SUNRISE ROOM, 1:00 TO 4:35 P.M.

Session R. Psychological and Physiological Acoustics I: Acoustics of Echolocating Animals in Air

Patricia Brown, Chairman

Department of Biology, University of California, Los Angeles, California 90024

Chairman's Introduction—1:00

Invited Papers

1:05

R1. Neural representation of bisonar information in the auditory cortex of the mustached bat. Nobuo Suga (Department of Biology, Washington University, St. Louis, MO 63130)

For echolocation, the mustached bat (*Pteronotus parnellii rubiginosus*) emits orientation sounds containing up to four harmonics, each of which consists of a long constant-frequency (CF) component followed by a short frequency-modulated (FM) component. The auditory system of this species extracts different types of bisonar information from complex orientation sound and echo pairs and represents these systematically in the separate areas of the cerebral cortex. Nine clusters of combination-sensitive neurons have thus far been found in the auditory cortex. Two of these clusters making up the CF/CF area consist of CF_1/CF_2 and CF_1/CF_3 facilitation neurons which are tuned to particular Doppler shifts, i.e., target velocities from 8 to -2 m/s. The remaining seven clusters consist of different types of FM-FM neurons which are tuned to particular echo delays, i.e., target distances. Three of these clusters, making up the FM-FM area, consist of FM_1-FM_2 , FM_1-FM_3 , and FM_2-FM_3 facilitation neurons. The FM-FM area projects bilaterally to the dorsal fringe (DF) and ventrolateral (VL) areas of the nonprimary auditory cortex. The DF area consists of three clusters of FM-FM facilitation neurons. The VL area contains a small cluster of facilitation neurons that are tuned only to the FM_1-FM_2 combination. The FM-FM area represents target distances from 7 to 310 cm, while the DF area represents these from 14 to 140 cm. [Work supported by 1-R01-NS 17333.]

1:50

R2. Fluttering target detection in horseshoe bats. Hans-Ulrich Schnitzler (Lehrstuhl Zoophysiology, Eberhard-Karls-Universitat, D-7400 Tuubingen, Federal Republic of Germany)

The rhythmical modulations in insect echoes caused by the moving wings are behaviorally relevant information for horseshoe bats. By evaluating this information the bats are able to detect insect echoes even in strong background clutter and probably to classify different species. Transmitter and receiver of the horseshoe bats' echolocation are especially adapted for the processing of this behaviorally relevant information. An adaptation of the transmitter is the Doppler shift compensation which keeps the carrier frequencies of the insect echoes within an "expectation window" and uncouples the insect echoes from the bats' flight movement. Adaptations of the receiver are the specialized structure of the cochlea with the highly expanded frequency representation in the range of the insect echo frequencies which leads to a strong overrepresentation of sharply tuned neurons with special response characteristics throughout the whole auditory pathway. Thereby the bats establish an "analysis window" in the receiver which corresponds to the "expectation window" caused by the transmitter characteristic.

Contributed Papers

2:35

R3. Echolocation and foraging behavior of *Hipposideros ruber* (Chiroptera). G. P. Bell and M. B. Fenton (Department of Biology, Carleton University, Ottawa, Ontario, Canada K1S 5B6)

Although a great deal of recent literature deals with the neurological and anatomical aspects of Doppler-shift compensation (DSC) by some bats, there are a few data on the foraging behavior of these species. We studied the foraging of *Hipposideros ruber* in Zimbabwe, and conclude from its behavior and echolocation calls that it is capable of DSC. These bats selected moths which were fluttering their wings whether the prey were flying or sitting on the substrate. If, during an attack, the moth stopped fluttering its wings, the bat terminated its approach. Pursuits of flying and sitting targets were equally successful, with the bats capturing the targets 40% of the time. Control experiments indicated that the bats did not use vision or the sound of fluttering to locate their prey. DSC appears to be a means of rejecting clutter. [Work supported by NSERC, Canada.]

2:50

R4. Echolocation behavior in a "flycatcher" bat, *Hipposideros diadema*. Patricia E. Brown and Robert D. Berry (Department of Biology, UCLA, Los Angeles, CA 90024 and Naval Weapons Center, China Lake, CA 93555)

Echolocation behavior and sonar pulse design in free-ranging Diadem horseshoe bats (*Hipposideros diadema*) were studied during the NSF-sponsored expedition to Chillagoe, Australia. To follow individual bats under field conditions, crystal-controlled radio transmitters were attached to the bats. This produced valuable data on home range, activity patterns, and foraging behavior as well as permitting the recording of echolocation signals of known bats under field conditions. In its foraging area, a bat would hang from a limb, acoustically scanning for a passing beetle. Upon detection, the bat briefly tracked its prey before darting out to capture it and returning to the same roost. *Hipposideros diadema* employs a CF/FM signal with a resting frequency of 55 kHz. They are capable of Doppler-shift compensation, altering the CF component in response to movements of nearby objects. While *Rhinolophus ferrumequinum* will lower its resting frequency to compensate for its own flight speed, it does not react to negative Doppler shifts caused by receding objects. *Hipposideros* is stationary for prey detection, and compensates for both negative and positive Doppler-shifted echoes generated by approaching and receding insects by lowering and raising the CF component of its sonar pulses.

3:05

R5. Mechanisms of sonar pulse production by oilbirds. Roderick A. Suthers and Dwight Hector (Medical Sciences Program, Indiana University, Bloomington, IN 47405)

The echolocating clicks of oilbirds (*Steatornis caripensis*) are generated in two bronchial semisyringes. Each click has a duration of 20 to 60 ms and may consist of a continuous broadband sound or be divided by a short silent interval into two shorter sounds. Electromyograms, together with measurements of subsyringeal pressure, tracheal pressure, and the rate of tracheal airflow, show that clicks are preceded by contraction of the ster-

notrachealis muscles which move the syrinx caudad, causing adduction of the syringeal membranes during expiratory airflow and initiating phonation. Subsequent contraction of a previously undescribed syringeal muscle terminates the click by rotating a syringeal cartilage and abducting the external tympaniform membranes. This syringeal muscle does not contract during longer duration vocalizations. It is thus specialized for the production of clicklike sounds. Simultaneous measurement of the rate of airflow through each semisyrinx indicates that during some clicks one semisyrinx closes while the other remains open. This is the first evidence that airflow through each side of the syrinx can be independently controlled during phonation. [Work supported by NSF.]

3:20

R6. Acuity of echolocation in the oilbird, *Steatornis caripensis*. David B. Thompson and Roderick A. Suthers (Medical Sciences Program, Indiana University, Bloomington, IN 47405)

The neotropical oilbird echolocates in totally dark caves with a click which has most of its energy between 0.5 and 3.0 kHz. The acuity of echolocation was measured with an array of stationary cylindrical obstacles (diameter 8.9, 3.2, or 0.5 cm) which were shifted between trials to prevent the birds from learning their position. Birds were tested in a dark room and observed with an infrared viewer. Flights through the row of obstacles were scored as hits, light touches, or misses. The birds' ability to negotiate the 3.2-cm array was significantly better than with 0.5-cm obstacles, indicating that oilbirds can acoustically detect 3.2-cm-diam obstacles. This is considerably smaller than the 20-cm threshold of previous studies [M. Konishi and E. I. Knudsen, *Science* **204**, 425-427 (1979)]. Like echolocating swiftlets [D. R. Griffin and D. Thompson, *Behav. Ecol. Sociobiol.* **10**, 119-123 (1982)] and bats, oilbirds can detect obstacles whose diameter is a fraction of the wavelength of their echolocating pulses. [Work supported by NSF.]

3:35

R7. Increased low-frequency sensitivity in the ears of Kauai moths: Response to the acoustic signals of the Hawaiian Hoary Bat. James H. Fullard [Department of Zoology, Erindale College, University of Toronto, Mississauga, Ontario, Canada L5L 1C6] and Jacqueline J. Bellwood (Department of Entomology, University of Florida, Gainesville, FL 42611)

The auditory sensitivity of tympanate moths from the island of Kauai was studied to examine the sensory effects placed on these insects by the site's sole bat species, *Lasiurus cinereus semotus* (Hawaiian Hoary Bat). Audiograms were neurologically determined and then subdivided into four bandwidths to render an estimation of sensitivity (audiogram area) at 5-25, 30-50, 55-75, and 80-110 kHz. Compared to moths analyzed from sites in Canada, Africa, and Papua New Guinea, Hawaiian moths reveal a reduced high-frequency (> 50 kHz) sensitivity coupled with a significantly higher sensitivity to frequencies below 30 kHz. Tuning is very broad with best frequencies extending from 9 to 33 kHz for various species. Free-flying individual bats were recorded as they haunted at the same site where moths were captured. *L. cinereus* emits two distinct signals as it flies: a typical FM, harmonically complemented, echolocation call centered at 27.9 kHz and another FM signal of 9.8 kHz, commonly emitted during agonistic encounters with other bats while foraging. We suggest

that this bat glean echolocation information from its low-frequency social call and thereby provides another set of acoustic frequencies to which sympatric moths have become sensitive. [Work supported by National Geographic Society and NSERC.]

3:50

R8. Responses of cerebellar units in FM and CF-FM bats to acoustic stimuli. Philip H.-S. Jen, Tsutomu Kamada, and Xinde Sun (Division of Biological Sciences, The University of Missouri, Columbia, MO 65211)

With pure tone pulses (35- and 0.5-ms rise-decay times), a total of 499 units from FM bats (*Myotis lucifugus* and *Eptesicus fuscus*) and 401 units from the CF-FM bats (*Pteronotus parnellii parnellii* and *Pteronotus parnellii rubiginosus*) were isolated from a large area of the cerebellar vermis and hemispheres. Response latencies ranged between 1.2 and 68 ms but most were below 10 ms. Phasic units, phasic bursters, and tonic units were observed. Threshold curves of these units were either narrow, broad, or irregular. In the CF-FM bats, units with BFs between 60 and 63 kHz corresponding to the predominant CF frequency were sharply tuned and showed off responses upon cessation of the stimulus. In both bats, variation of BFs of isolated units from penetrations within the same lobe was smaller than those from different lobes. A study of frequency tuning and representation suggests that a bat's cerebellum can effectively process the predominant components of biologically significant signals. Responses to a 4 ms upward and downward sweeping FM stimuli and pure tone pulses were studied in 70 units of the *Eptesicus fuscus*. Most units were more sensitive to a downward sweeping FM stimulus than to a pure tone pulse. An upward sweeping FM stimulus was as effective as a 35-ms pure tone pulse. A study of directional sensitivity of 47 units showed that 33 units were most sensitive to a sound delivered from the front. The other 14 units were most sensitive to a sound delivered from 20°-30° lateral. [Work supported by NSF 80 07348 and USPH 1-K04-NS-00433 to P. Jen.]

4:05

R9. A new method for measuring head aim in echolocating bats. W. Mitch Masters and James A. Simmons (Institute of Neuroscience, University of Oregon, Eugene, OR 97403)

A new method has been developed for continuously monitoring the direction in which an echolocating bat is pointing its head. Two light emitting diodes are mounted on the bat's head and their positions read sequentially, under microprocessor control, using an x-y position sensing photodiode. The microprocessor calculates the angle of the line between the two light emitting diodes with respect to an initial direction, and the location of the midpoint of this line. Readings can be made over a 360° range with an accuracy of $\pm 1^\circ$, at a rate of 100/s. This device has been used to determine the bat's head aim in measurements of horizontal angular resolution. Simultaneous monitoring of the bat's sonar emissions provides data on how information is integrated over time by the bat. Results of these studies will be discussed. [Work supported by NIH and NSF.]

4:20

R10. Spectrogram processing in echolocation. Richard A. Altes (ORINCON Corporation, 3366 N. Torrey Pines Court, La Jolla, CA 92037)

Echolocation behavior and neuronal representations of sound suggest that animal sonar can be modeled in terms of a spectrogram correlation process. Recent experimental results, however, indicate that bats may be sensitive to the fine structure of the signal-echo cross-correlation function (J. A. Simmons, *Science* **204**, 1336). This sensitivity cannot be explained by cross correlation of spectrograms. Spectrogram deconvolution or an analysis-by-synthesis procedure can be used to reconstruct a signal from its spectrogram, aside from an arbitrary phase shift. Cross correlation of two such reconstructed signals, with the same arbitrary phase shift assigned to both, yields the required sensitivity to cross-correlation fine structure. The waveform reconstruction procedure is accurate only if spectrogram filters are not smooth window functions as in man-made processors, but have large time-bandwidth product, e.g., sharp rise and slow decay in the frequency domain, as in critical band models of hearing. This filter property insures that the filter ambiguity function has the same or larger support in the range-Doppler plane as the signal ambiguity function, a necessary condition for spectrogram deconvolution.

TUESDAY AFTERNOON, 8 NOVEMBER 1983

GOLDEN WEST ROOM, 1:00 TO 4:25 P.M.

Session S. Psychological Acoustics II: Masking and Noise Discrimination

Dominic W. Massaro, Chairman

Department of Psychology, University of California, Santa Cruz, California 95064

Chairman's Introduction—1:00

Contributed Papers

1:05

S1. Stimulus parameters governing confusion effects in forward masking. Donna L. Neff^{a)} and Daniel L. Weber (Boys Town National Institute for Communication Disorders in Children, 555 N. 30th Street, Omaha, NE 68131)

Confusion effects in forward masking occur when the listener has difficulty discriminating a suprathreshold signal from the preceding masker. Confusion effects were examined for "pulsing" forward maskers composed of repeated bursts of a sinusoid followed by a sinusoidal signal. Differences between the level, frequency, and duration of the signal and an individual masker pulse, as well as offset-onset delay were varied to determine the change necessary to resolve confusion. For maskers composed of 20-ms pulses, changes in signal level of 1-5 dB or in signal fre-

quency or 20-30 Hz resolved confusion. For maskers composed of 10-, 20-, or 40-ms pulses, the signal delay necessary to resolve confusion increased with masker-pulse duration but was typically less than 10 ms. For these three pulsing maskers, signal durations less than one half or greater than twice the masker-pulse duration were necessary to resolve confusion. [Work supported by NSF, NIH, and the University of Nebraska.]^{a)} Present address: Laboratory of Psychophysics, Harvard University, 33 Kirkland Street, Cambridge, MA 02138.

1:25

S2. The occlusion-impedance of the ear canal and its influence on the occlusion effect and external physiological masking. Christoph Pössel (Lehrstuhl für allgemeine Elektrotechnik und Akustik, Ruhr-Universität Bochum, Postfach 102148, West Germany)

Bone conduction (BC) as the limiting factor for the sound protection rate of earplugs and earmuffs can be calculated by models. A model of this kind is part of a dummy head system, recently developed in our laboratory for objective sound attenuation measurement of sound protection devices. Based on the separation of BC into external-, middle-, and inner-ear components, the model allows determination of BC according to the actual type of occlusion. BC is mainly affected by the acoustic impedance, seen outwards from the entrance to the ear canal. We have measured this "occlusion-impedance" for different occluding devices. Sample results will be presented. The occlusion effect is, then, determined by the occlusion impedance and, also, by the relative vibrations between skull and earmuff, the latter depending on the mechanical impedance of earmuff and underlying skin as well as on the vibration modes of the skull. Further, the physiological noise spectrum, as generated in the external ear, is affected by the occlusion impedance too, thus leading to a characteristic external physiological masking, former known as the "missing 6-dB effect." [Work partly supported by Minister für Wissenschaft und Forschung, NRW.]

1:45

S3. Frequency discrimination of tones presented either inside or outside of "notches" in band-reject noise. David S. Emmerich, Deborah A. Fantini, and William S. Brown (Department of Psychology, State University of New York at Stony Brook, Stony Brook, NY 11794)

At the 104th meeting of this Society we reported data consistent with the notion that, in frequency discrimination, information is integrated over a wide frequency range. These data were obtained from subjects who were asked to discriminate tones which were presented in the centers of "notches" of various widths in backgrounds of band-reject noise. The present research extends the earlier findings by presenting the tones close to the edges of bands of rejected frequencies in an attempt to determine the relative importance of activity from frequency regions higher and lower than the nominal frequencies of the tones being discriminated. In addition, control experiments were conducted in which the tones were presented in the passbands of band-reject noise rather than inside the notches. Again the results suggest that activity remote from the peak of the distribution of activity created by the presentation of a tone is important in frequency discrimination. A reasonable interpretation of the data is that remote effects extend more widely on the high-frequency side than on the low-frequency side of the tones to be discriminated. This interpretation is somewhat clouded by the unexpected finding that in some conditions adding to the spectral content of the noise background improved rather than impaired performance.

2:05

S4. On the form of the masking function for intensity discrimination of pure-tones. Neal F. Viemeister and Sid P. Bacon (Department of Psychology, University of Minnesota, Minneapolis, MN 55455)

The published data on intensity discrimination of tones does not permit detailed characterization of how the difference threshold (ΔI) varies with intensity (I) over a wide intensity range. In the present study, masking functions (ΔI vs I) were obtained for 200-ms, 1-kHz tone bursts over a 115-dB range with I varied in 2.5- or 5-dB steps. For intensities between 5-dB SL and 95-dB SPL, the masking functions were in reasonable agreement with recent reports: the Weber fraction, $\Delta I/I$, was approximately constant up to 40 dB SPL and then decreased monotonically up to 95 dB SPL. For I near absolute threshold, some subjects showed a slight "sensitization," in which ΔI was below absolute threshold. Above 100 dB SPL, the Weber fraction increased monotonically with increasing intensity, suggesting mild "saturation." These data should permit more refined modeling of intensity discrimination of tones and of certain complex waveforms. [Supported by NINCDS NS12125.]

2:25

S5. Forward masking tuning curves obtained in the presence of a backward noise masker. Brian R. Shelton and John C. Booth (Departments of Psychology and Communicative Disorders, University of Western Ontario, London, Ontario, Canada N6A 5C2)

The combined effect of a backward and forward mask has often been noted to be difficult to predict on the basis of the masking obtained from either masker alone. This report describes the nonlinear effects of a backward mask on the shape of forward masking tuning curves. Five-point psychophysical tuning curves were obtained in two observers, using 200-ms forward maskers of 500, 750, 1000, 1250, and 1500 Hz, and a 10-ms, 1000-Hz signal presented at 40 dB SPL. The forward mask had a 2-ms rise/fall time and the signal was fast-gated 3 ms after the forward mask. The backward mask was a 100-ms burst of white noise, presented 3 ms after the signal offset, with a fast rise/fall. Tuning curves were obtained in quiet, and with backward maskers of 20, 50, and 80 dB SPL. Tuning curves became sharper as the level of the backward mask was increased. The tip of the tuning curve dropped by at least 20 dB in the presence of the 80-dB backward mask, and by less than 10 dB at other frequencies. The results are discussed in terms of the strategies employed by listeners to detect a signal in the presence of a forward mask. [Work supported by NSERC.]

2:45

S6. Detection cues in forward masking and their relationship to off-frequency listening. William S. Yacullo (Section of Communicative Disorders, Rush-Presbyterian-St. Luke's Medical Center, Chicago, IL 60612) and Paul J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Previous studies [D. Johnson-Davies and R. D. Patterson, *J. Acoust. Soc. Am.* 65, 765-770 (1979) and B. J. O'Loughlin and B. C. J. Moore, *J. Acoust. Soc. Am.* 69, 1119-1125 (1981)] suggest that off-frequency listening is a major factor contributing to sharpness of psychophysical tuning curves. The present study is designed to evaluate the contribution of off-frequency listening to the sharpness of psychophysical tuning curves as a function of temporal characteristics of both the variable masker and stationary masker used to restrict listening to a narrow region surrounding the probe using the restricted-listening tuning curve paradigm first employed by Johnson-Davies and Patterson (1979). Conventional forward-masked psychophysical tuning curves initially were generated using stimuli of varying temporal characteristics (sinusoids, AM and QFM tones, and AM noise) as the variable masker. Restricted-listening tuning curves subsequently were obtained utilizing fixed-level stationary maskers 100 Hz wide of differing temporal characteristics (QFM tone, AM noise, and synthetic noise comprised of sinusoids of equal amplitude, random phase relation, and spaced 1 Hz apart) which were gated with the variable masker. Results suggest that the off-frequency listening effect observed utilizing a tuning curve paradigm is largely dependent on the temporal characteristics of both variable and stationary maskers. Assuming differing detection mechanisms for maskers of varying temporal characteristics, it appears that the different detection cues are not equally dependent on off-frequency listening.

3:05

S7. Masking ability of forward maskers of varying temporal characteristics. William S. Yacullo (Section of Communicative Disorders, Rush-Presbyterian-St. Luke's Medical Center, Chicago, IL 60612) and Paul J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

It has been suggested that the detection cues for simultaneously presented sinusoidal and noise maskers are different [D. Green, *J. Acoust. Soc. Am.* 41, 1517-1525 (1967) and D. Weber and R. Patterson, *J. Acoust. Soc. Am. Suppl.* 1 68, S37 (1980)]. The present study was designed to evaluate masking ability of forward maskers of varying temporal characteristics. Conventional forward-masked tuning curves were generated utilizing a sinusoidal probe and four types of stimuli for the masker: a sinusoid, quasi-frequency-modulated (QFM) tones, AM tones, and AM narrow-band noise 100 Hz wide. These maskers were selected as their envelopes possess a wide range of amplitude fluctuation. The AM and QFM tones (50 Hz wide) had identical spectral structures, yet distinctly different temporal characteristics. Differences in masker effectiveness were evaluated by comparing the tuning curves generated using the various types of maskers. Results indicate that stimuli with amplitude fluctuations that are large and random in nature (AM noise) exhibit greater

forward masking ability that stimuli with either minimal or no envelope variation (QFM tone and sinusoid) or large, yet periodic, amplitude perturbations (AM tone). This difference in masking ability is exhibited on the high-frequency branch and within the tip region of the tuning curve. It is suggested that differences in masking ability may reflect the use of different signal detection cues.

3:25

S8. Improved 1-bark bandwidth auditory filters. Andrew Sekey and Brian A. Hanson (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

A critical band filtering function attributed to Schroeder [A. J. Fourcin, Rapporteur, "Speech processing by man and machine," in Theodore H. Bullock (1977), *Recognition of Complex Acoustic Signals*, Dahlem Konferenzen, Berlin] is modified so as to (i) ensure 1-bark bandwidth, (ii) have asymptotic slopes which reflect masking characteristics of the human ear, and (iii) attains its maximum at 0 bark. A bank of the new filters spaced at 1-bark intervals is presented, and its smoothing effect on a vowel spectrum is illustrated.

3:45

S9. Discrimination of noise as a function of bandwidth and duration. Thomas E. Hanna (Laboratory of Psychophysics, Harvard University, 33 Kirkland Street, Cambridge, MA 02138)

A same-different task was used to study the discriminability of noise bursts as a function of duration (0.1, 0.4, 1.6, 6.4, 25.6, 102.4, or 409.6 ms) and bandwidth (50, 200, 800, or 3200 Hz) of the burst. Each trial contained either identical (same) or independent (different) samples of noise,

with the noise bursts sampled randomly for each trial. Performance improved as a function of bandwidth, as would be expected based on the increase in information in the stimulus. However, performance changed nonmonotonically as a function of duration, even though there is a similar increase in information as duration increases. For each of the four bandwidths, performance improved up to a duration of 25.6 ms, but declined with further increases in duration. Subsequent experiments were conducted to explain the decrease in performance for longer durations. The results will be discussed in terms of the peripheral and central determinants of performance for this task. [Work supported by NIH.]

4:05

S10. Changes in differential sensitivity for frequency modulation in noise-exposed chinchillas. William W. Clark, Barbara A. Bohne, and Glenis R. Long (Central Institute for the Deaf, St. Louis, MO 63110)

Frequency modulation discrimination thresholds were obtained for two chinchillas before, during, and after exposure to an octave band of noise centered at 0.5 kHz, 95 dB SPL, for 54 days. During exposure, when the animals had threshold shifts of 35–45 dB, difference limens for frequency modulation (DLs) were impaired by ten-fold for center frequencies below 2.3 kHz; at 2.3 kHz and above they were completely unaffected. Thirty days after exposure, when threshold shifts had recovered to about 25 dB, DLs recovered about halfway (fivefold reduction) for the low-frequency stimuli; high-frequency DLs remained normal. Histological evaluation of the cochlea from these two animals showed losses of outer hair cells in the apical turn averaging 30%, and a basal-turn lesion in one animal at a point 60% of the distance from the apex. The results of this study extend our previous observation that the base and the apex of the cochlea are affected differently by exposure to noise. [Work supported by NINCDS.]

11:05

T5. Nearfield acoustical holographic techniques used to visualize radiated sound fields. J. D. Maynard (Department of Physics, The Pennsylvania State University, University Park, PA 16802)

The term signal processing is customarily used in reference to the analysis of data obtained as a function of time; similarly the term image processing refers to the enhancement of two-dimensional spatial data. For wave fields, such as optical or acoustical fields, the wave equation couples the space and time variables and permits an extremely powerful signal processing technique referred to here as generalized holography. In this technique temporal data are measured over a two-dimensional spatial surface, and the data are processed to reconstruct and visualize the entire three-dimensional wave field. The power of this technique arises from the enormous expansion of information which occurs when going from the two-dimensional measurement to the three-dimensional reconstruction. Although the theory of generalized holography is exact, conventional optical and acoustical holography suffers from serious intrinsic limitations, such as the spatial resolution being limited to the radiated wavelength. At The Pennsylvania State University we are developing a new acoustic generalized holographic technique, called Nearfield Holography, which overcomes the limitations of conventional holography. In this technique a brief measurement with a microphone array can be used to visualize the surface motion of a complex vibrating structure, to reconstruct the entire three-dimensional sound pressure field, particle velocity field, and vector intensity field, and to map the flow of acoustic energy from the sources to the farfield. With this system one may pinpoint (within centimeters) acoustic energy producing sources on vibrators with small structural features (such as musical instruments) even when the radiated wavelength is several meters. In this talk our most recent Nearfield Holography equipment and processing algorithms will be described, and computer graphic reconstructions of vibrating surfaces (moving in real-time) and radiated sound fields will be shown. [Nearfield holography research is supported by the Office of Naval Research (Physics Program) and the National Aeronautics and Space Administration.]

11:35

T6. Signal processing techniques used in underwater acoustics. V. C. Anderson (Marine Physical Laboratory of the Scripps Institution of Oceanography, University of California, San Diego, CA 92152)

The two underlying goals of most signal processing applications in underwater acoustics are detection and estimation of a signal in the presence of uncertainties. The use of standard techniques such as matched filtering, prewhitening, and adaptive parameter estimation brings into play new dimensions of technology in the face of the wide variability and uncertainty in parameters of the signal and the background. This paper will present an overview of signal processing applications in the field.

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

CABINET ROOM, 1:30 P.M.

Meeting of Standards Committee S2: Mechanical Shock and Vibration

to be held jointly with the

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

P. H. Maedel, Jr. Chairman S2

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

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

220 Clark Avenue, Branford, Connecticut 06405

Standards Committee S2 on Mechanical Shock and Vibration. Working group chairs will present reports of their recent progress on writing and processing various shock and vibration standards. There will be a report on the interfaces 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). Report on the ISO/TC 108 meeting, held from 19-30 September 1983.

Session U. Psychological and Physiological Acoustics II (Poster Session)

Patrick W. B. Moore, Chairman

Code 512, Naval Ocean Systems Center, Kailua, Hawaii 96734

Chairman's Introduction—1:00

All posters will be displayed from 1:00–3:00 P.M. To allow all contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00–2:00 P.M. and contributors of even-numbered papers will be at their posters from 2:00–3:00 P.M.

U1. The effect of contralateral stimulation on spontaneous acoustic emissions. John H. Grose (Auditory Research Laboratory (Audiology), Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

Subjects exhibiting spontaneous acoustic emissions (SAEs) were presented with wideband noise in the ear contralateral to that in which their SAE was being recorded. The SAE magnitudes were monitored while the noise was incremented in 5-dB steps. Some SAE components were markedly reduced with increasing contralateral stimulation while others appeared to be enhanced, despite frequency separations between components of less than 200 Hz. The effects were in the order of 3–10 dB, though occasionally greater, and the threshold for the SAE change was generally below the acoustic reflex threshold for that subject. The rms level of the ear canal noise in the ipsilateral ear was also monitored to ensure that none of the observations were a result of transcranial transmission of the stimulus. Pure tones were substituted for the noise in order to determine whether the effect was tuned; however no sharp tuning was observed.

U2. The perception of individual "harmonics" of a ripple-noise spectrum. E. M. Burns, M. Corban (Purdue University, West Lafayette, IN 47907), L. L. Feth, and V. Kirby (University of Kansas, Lawrence, KS 66045)

A series of experiments were performed in order to compare the ability of listeners to obtain pitch information from a single "harmonic" (ripple) of a ripple-noise spectrum with their ability to obtain similar information from an equivalently filtered white noise spectrum. The experiments included: (1) a study to determine the ability of listeners to "hear out" individual ripples from a ripple noise using Plomp's paradigm [Plomp, *J. Acoust. Soc. Am.* 36, 1628–1636 (1964)], (2) the determination of "frequency" JNDs for individual ripples and for narrow-band noises, (3) the determination of "fundamental frequency" JNDs (by randomizing ripples within trials), and (4) musical-interval-identification studies. The results are discussed in relation to temporally based vis a vis spectrally based models of ripple noise pitch perception. [Supported by NINCDS.]

U3. The evaluation of the effectiveness of three hearing protection devices at an industrial facility with a daily TWA of approximately 107 dB. Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27650) and Julia Doswell Royster (Environmental Noise Consultants, Inc., Cary, NC 27511)

Three hearing protection devices (Norton Sigma Comfit earplug, Defeco Silenta earmuff, and E-A-R foam earplug) were compared for their effectiveness in preventing both daily temporary threshold shifts (TTS) and shifts between sequential annual audiograms (%BW statistic) [L. H. Royster and J. D. Royster, in *Personal Hearing Protection in Industry* (Raven, New York, 1982)]. The hearing protection devices varied significantly in preventing TTS. Comfit wearers exhibited the largest TTS shifts, which differed from zero at 500–4000 Hz. Significant TTS was found for Silenta wearers at 500 Hz, and for E-A-R wearers at 1000–2000 Hz. E-A-R wearers showed nonsignificant improvements at 3000–6000 Hz, and Silenta wearers showed significant improvement at 6000 Hz. Employees' previous three annual audiograms were evaluated to compare %BW values for the wearer groups. These values were 53% for Comfit wearers,

45% for Silenta wearers, and 26% for E-A-R wearers. Previous research indicates that the %BW statistic for a protected or non-noise population with past audiometric test experience will be less than 30%.

U4. A new approach to auditory depth perception. R. W. Gatehouse (Department of Psychology, University of Guelph, Guelph, Ontario, Canada N1G 2W1)

Acoustic depth judgment studies have generally used static position sources, and asked subjects to estimate the distance of one source only, or one source relative to a second. Here, blindfolded subjects were asked to "align" a moveable medial sagittal plane comparison source with a static 1.0-kHz standard, 3 m away from them. The comparison source's starting position could be at 0 (position of standard), ± 100 or ± 200 cm away and was either a 0.7- or 1.3-kHz narrow-banded tone. The two sources alternated in 3-s bursts till the S indicated "alignment" was achieved. Half the subjects received the 1.0 and 0.7 kHz pairing first then the 1.0 and 1.3 kHz pairing and vice versa. Three trials were given for each pairing from each starting position. The mean error scores (i.e., mean differences between "true" and estimated source position) were analyzed. The results indicate that observers overestimate the distance of the standard when initial position of the comparison is close to them, and underestimate it when the comparison starts from beyond the "0." Comparison frequencies performances did not differ. The results partially confirm Mershon and Bowers [Percept. 8, 311–322 (1975)] using a more standard depth perception paradigm.

U5. Listeners' identification of human-imitated animal sounds. Norman J. Lass, Sandra K. Eastham, Tammie L. Wright, Audrey R. Hinzman, Karen J. Mills, and Amy L. Hefferin (Department of Speech Pathology and Audiology, 805 Allen Hall, West Virginia University, P. O. Box 6122, Morgantown, WV 26506-6122)

The purpose of this investigation was to determine if listeners can accurately identify human-imitated animal sounds. A total of 20 speakers, ten females and ten males, recorded their imitations of cows, cats, dogs, pigs, and sheep. A master tape containing the randomly arranged recordings was prepared and presented to 30 judges who were asked to identify the imitated animals and to rate the confidence of their judgments on a seven-point rating scale. Results indicate that: (1) listeners' accuracy was relatively high for all five human-imitated animals; (2) imitations of dogs were most accurately identified, followed by sheep, cows, cats, and pigs; (3) for all five animals investigated, listeners' accuracy was greater for female-imitated than male-imitated sounds; (4) listeners' confidence ratings were highest for sheep followed by cows, dogs, cats, and pigs; and (5) listeners' confidence ratings were higher for female-imitated than male-imitated animal sounds (with the exception of dog sounds). Implications of these findings and suggestions for future research are discussed.

U6. Effects of signal starting phase on diotic and dichotic detection in reproducible noise maskers. R. H. Gilkey and B. Kollmeier¹⁾ (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Dolan, Hirsh, and Yost [*J. Acoust. Soc. Am.* 70, 886–887 (1981)] and Gilkey, Hanna, and Robinson [*J. Acoust. Soc. Am. Suppl.* 1 68, S59

(1980)] have shown that the detectability of a tonal signal varies greatly as a function of the phase relationship between the signal and a reproducible noise masker under diotic (NoSo) presentation. However, detectability varies only slightly, if at all, as a function of phase under dichotic (NoS π) conditions. In the present experiment an adaptive two-alternative forced-choice procedure is used to obtain detection thresholds for a 500-Hz tonal signal in the presence of each of six individual samples of reproducible noise. Six binaural conditions (NoSo, NuSo, N π Sm, NoSm, N π So, and NoS π) were investigated in combination with six signal starting phases (0°, 60°, 120°, 180°, 240°, and 300°). The results indicate that although the magnitude of the phase effect decreases with larger masking level differences, it is similar in form for all conditions with the same masker. [Work supported by NINCDS Grant NS 03856, and the Fulbright Commission, Federal Republic of Germany.]^{a1} Also at Drittes Physikalisches Institut, University of Goettingen, Federal Republic of Germany.

U7. Perceived rate of randomly modulated sounds. Hisashi Kado (Central Institute for the Deaf, St. Louis, MO 63110 and Electrotechnical Laboratory, Sakura, Ibaraki 305, Japan)

The common measures used in evaluating the magnitude of a noise (L_x , L_{eq} , and etc.) correspond to first- and second-order statistics of the distribution of sound levels. There are, however, perceived differences between sounds that have the same values of these statistics. It is therefore necessary to consider the spectral properties of the waveform envelope which correspond to higher-order statistics. In the present experiments the subject compares a test stimulus and a reference stimulus and selects the one which appears to be fluctuating more rapidly. The reference stimulus is a noise whose waveform envelope is a noise with a low-pass cutoff which varies from 0.125 to 8 Hz. The point of subjective equality is determined by adjusting the modulation frequency of the test stimulus, a sinusoidally modulated noise, in an adaptive procedure. The results indicate that when the median frequency of the spectrum of the envelope is above 1 Hz, it is a reasonably good predictor of the subjective rate, when the median is below 1 Hz the mode is a better predictor. [Work supported by NINCDS.]

U8. Effect of presentation level on the SSW test with hearing-impaired adults. Patricia A. Flynn, Jeffrey L. Danhauer, Dennis J. Arnst, Monica C. Goller, and Sanford E. Gerber (Speech Department, University of California, Santa Barbara, CA 93106)

The effects of varying presentation level on the Staggered Spondaic Word (SSW) test scores of sensorineural hearing-impaired subjects were investigated. Subjects were 15 adults having cochlear hearing losses between 30 and 50 dB (*re*: PTA). Performance-intensity (PI) functions were generated using subjects SSW test scores. Each subject received all 40-SSW test items, but the PI functions were generated by presenting items numbered 1-10 at 20 dB SL (*re*: three frequency PTA); 11-20 at 30 dB SL; 21-30 at 40 dB SL; and 31-40 at 50 dB SL. These results for the hearing-impaired were compared to functions reported earlier for normal subjects [P. C. Doyle, thesis, University of California, Santa Barbara (1981)]. The PI functions were analyzed for "competing" versus "non-competing" conditions, and differences between subjects' raw SSW scores and word discrimination scores as a function of presentation level. Practical and theoretical implications for the clinical presentation of the SSW test at the various intensity levels to hearing-impaired adults are discussed.

U9. A comparison of the temporal response of the auditory system under dichotic and diotic conditions. B. Kollmeier^{a1} and R. H. Gilkey (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Most of the classical experiments on binaural masking have held the interaural phase of the noise, as well as that of the signal, constant throughout the observation interval. In this experiment the detectability of a 500-Hz tonal signal is investigated in the presence of a 2000-Hz low-pass noise, which is switched from N π to No at the midpoint of the observation interval. For a 20-ms S π signal there is approximately a 15 dB increase in detectability when it is presented late in the interval, relative to early in the interval. The time course of the change in detectability can be used as an indication the decay of activity in the binaural system. This

decay of activity is compared to a similar diotic configuration in which the level of the noise is decreased by 15 dB. The results indicate that the decay of activity in the binaural system is more "sluggish" than the decay of activity in the monaural system. [Work supported by NINCDS Grant NS. 03856, and the Fulbright Commission, Federal Republic of Germany.]^{a1} Also Drittes Physikalisches Institut, University of Goettingen, Federal Republic of Germany.

U10. Predicting masking by combinations of simultaneous maskers. Robert A. Lutfi (Auditory Research Laboratory, Northwestern University, Evanston, IL 60201)

In a previous paper [Lutfi, *J. Acoust. Soc. Am.* **73**, 262-267 (1983)], I proposed the following rule for predicting masking by combinations of simultaneous maskers; $X_c = [X_1^p + X_2^p + \dots + X_n^p]^{1/p}$, where X_1, X_2, \dots, X_n are the individual masking effects of the maskers, X_c is the combined effect, and $0.20 < p < 0.33$. In this paper, the rule is used to predict the results of several experiments in which the individual effects of maskers in the combination are known [Bilger, *J. Acoust. Soc. Am.* **31**, 1107-1109 (1959); Green, *J. Acoust. Soc. Am.* **41**, 1517-1525 (1967); Canahl, *J. Acoust. Soc. Am.* **50**, 471-474 (1971); Patterson and Nimmo-Smith, *J. Acoust. Soc. Am.* **67**, 229-245 (1980)]. With the possible exception of the study by Bilger, a single value of p ($= 0.3$) yields predictions in good agreement with the data. The rule also predicts some general characteristics of the results of studies in which the individual effects of the maskers are not known but can be estimated [Johnson-Davies and Patterson, *J. Acoust. Soc. Am.* **65**, 765-770 (1979); O'Loughlin and Moore, *J. Acoust. Soc. Am.* **69**, 1119-1125 (1981)]. [Work supported by NSF and NIH.]

U11. The effect of signal SPL on interaural time discrimination. Marion F. Cohen, Anne E. McClave, and Patricia Gregorio Pollanck (Department of Communication Sciences, The University of Connecticut, Storrs, CT 06268)

Interaural time jnds for a partially masked sinusoidal signal were measured as a function of signal SPL, with the signal presented at a constant signal-to-noise ratio of approximately 10 dB above its masked detection threshold. The 250-, 500-, or 1000-Hz signal ranged in level from 13 to 58 dB SPL. The masker was a continuous white noise. A block up-down two-interval forced-choice procedure was used. Results show that interaural time discrimination improves with increasing signal SPL despite the constant S/N ratio, indicating that signal SPL is a primary determinant of precision of interaural time discrimination. When these data are compared with jnds measured at the same signal SPLs but with no masker, they are nearly identical at 500 Hz. At 250 and 1000 Hz they are nearly identical at the higher signal levels for two of the three subjects. At the lower signal levels, the jnds measured in the presence of the masker are somewhat larger, but demonstrate a similar decrease with increasing signal level. The differences noted across frequency may be due to the effect of the internal noise. [Work supported by NIH Grant No. 5 R01 NS16802.]

U12. Response durations and false alarm reaction times. John L. Orr (Department of Physiology, University of Texas Health Science Center, Dallas, TX 75235)

Three Long-Evans rats were initially trained to depress and hold down a response lever in the presence wideband white noise pulsing at 6.7 Hz for a variable period between 4 and 6 s to obtain a 45-mg food pellet. This reinforcement schedule is the same as operating a one-lever press-release psychophysical procedure with a variable foreperiod of 2-4 s and a 2-s trial duration with only no-signal trials. The initial training of the nonsignal trial correct response of holding the lever down throughout the trial permits the measurement of the temporal characteristics of the background level of release responses in the absence of a history of contingent reinforcement for releasing the lever. After training to asymptotic performance on the nonsignal trials alone, 50% signal trials (where the noise pulsation rate changed to 12.5 Hz) were included. Discrimination training was followed by changes in response duration histograms and false alarm reaction time histograms. In two of the three rats, there was an increase in the probability of incorrect lever releases 3-6 s after the lever was de-

pressed. [Work supported by NIEHS grant ROIES02750 to R. M. Lebovitz.]

U13. Auditory evoked potential asymmetries in right- and left-handed listeners. L. J. Hood, D. A. Martin, and C. I. Berlin (Department of Otorhinolaryngology and Biocommunication, Kresge Hearing Research Laboratory, Louisiana State University Medical Center, 1100 Florida Avenue, New Orleans, LA 70119)

While Decker [ASHA 23, 803 (1981)] reported no relationship of handedness to auditory tract preference in the auditory brainstem response, we know of no data for the middle and late potentials. This study examines the relationship of binaural interaction and lateral asymmetry in the middle and late potentials to handedness. Binaural, right monaural, and left monaural 100- μ s pulses were presented to six right-handed and six left-handed normal-hearing subjects in four conditions of earphone and electronics rotations. Middle and late potential data were analyzed statistically using a time series ANOVA. Lateral asymmetries and binaural interaction were derived by subtracting left from right responses and summed monaural from binaural responses, respectively. Data indicate individual differences in the amount and direction of asymmetry. The characteristics of binaural interaction and lateral asymmetry, the relationship to handedness, and the influence of stimulus and recording asymmetries will be described for the middle latency and late potentials. [Supported by NINCDS #NS11647, the Louisiana Lions Eye Foundation and the Eye and Ear Foundation.]

U14. Crib-o-gram (COG) and ABR effect of variables on test results. Laura B. Wright and Leonard P. Rybak (Department of Surgery, Southern Illinois University School of Medicine, Springfield, IL 62702)

This study compared seven categories of apparently relevant information concerning the Crib-o-gram (COG) as screening test and evaluated the following variables: birth weight, gestational age at birth, Apgar scores at 1 and 5 min, apnea, ototoxic drugs, number of days on assisted ventilation, and gentamicin blood levels. Low birth weight, early gestational age, and apnea were significantly correlated with COG results. Thirteen of the 55 newborns tested failed the COG at least twice before having the ABR test given. Twelve infants had normal ABR and one had some degree of hearing loss. Consequently the rate of false positive finding is very high. In spite of the high risk factors of the newborns, gentamicin, Apgar scores, or number of days on a ventilator were not implicated as a cause of hearing loss. [Work supported by Deafness Research Foundation.]

U15. Effects of cochlear high-frequency hearing loss on BSER. Neil T. Shepard (Otolaryngology Research Laboratories, Henry Ford Hospital, 2799 W. Grand Boulevard, Detroit, MI 48202) and John C. Webster (22250 Providence Drive, Suite 701, Southfield, MI 48075)

The extensive use of Brainstem Evoked Response Testing (BSER) for the detection of mass lesions involving the VIIIth nerve necessitates knowledge of the effects of hearing loss of cochlear origin on BSER. Thirty-one ears displaying varying degrees of sensorineural, high-frequency (beginning above 500 Hz) loss of sensitivity of cochlear origin were tested with BSER. A 100- μ s rectangular pulse was used to excite TDH-39 earphones at a rate of 11.3 pulses/s. Utilizing monaural stimulation at two to five intensities per ear, responses were obtained from a vertex to ipsilateral mastoid electrode montage. Linear regression analysis lines for waves I, III, and V latencies versus average degree of loss for various combinations of audiometric test frequencies are shown. The dependence of BSER wave latencies on individual and audiometric test frequencies and various combinations at and above 1 kHz are shown via the use of correlation coefficients. The results compare well with earlier results [A. C. Coats and J. L. Martin, Arch. Otolaryngol. 103, 605-622 (1977)] and imply the dependence of results on click stimulus parameters and earphone spectral characteristics. Also suggestions for normative limits of wave V latency shift for varying degrees of high-frequency cochlear hearing loss are given.

U16. Auditory brainstem response signal transduction through a Telephonics TDH-39 earphone. C. Charles Olsen (Department of Speech and Hearing Sciences, Texas Tech. University, Lubbock, TX 79410)

Waveforms and power spectra of acoustic signals used to elicit the auditory brainstem response were analyzed with respect to electrical and acoustical differences. Signals were of alternating polarity and included a dc pulse, tone bursts of 0.5 and 4.0 kHz, a 4.0-kHz tone pip, and a 2.0-kHz logon. The earphone faithfully transduced the temporal dimension of the 5.0-ms bursts and pips while the logon (450 μ s) and the click (100 μ s) durations were on the order of ten times greater than the electrical input. Earphone distortion included the loss of well-defined energy peaks and a spread of energy to higher frequencies. Signal polarity was not maintained through the earphone.

U17. The electrically induced auditory brainstem response in the guinea pig. D. D. Brown and R. T. Miyamoto (Department of Otolaryngology-Head and Neck Surgery, Riley Hospital A56, Indiana University School of Medicine, Indianapolis, IN 46223)

The cochlear implant is a clinically useful prosthesis for selected deaf patients who cannot benefit from a hearing aid. However, the poor speech discrimination and limited dynamic range generally achieved with the implant indicate that there are a number of basic and applied questions to be answered. Using acoustically and electrically elicited Auditory Brainstem Response (ABR) techniques, we have established an animal model (guinea pig) to help us answer some of these questions. We are presently studying the effect of electrode location on the electrical ABR. The single active cochlear electrode is located in either the basal or apical turn. The return electrode locations are the temporalis muscle, the eustachian tube, or the bulla. By choosing the proper polarity of the biphasic stimulating pulse, the electrical artifact can be minimized and virtually all of the ABR can be observed. Normative data for the electrical ABR and preliminary results of the electrode placement study will be presented.

U18. Complete reconstruction of the auditory system: Adult gerbil. David M. Harris and Michael Timmins (Department of Otolaryngology-Head and Neck Surgery, University of Illinois at Chicago, College of Medicine, 1855 West Taylor Street, Chicago, IL 60612)

The object of our work is the reconstruction of the entire auditory system of the adult gerbil. The methods to be described include perfusion with a vital stain, fixation and decalcification, photography of the cut surface of the tissue block (entire head) during collection of serial sections, tracing and digitizing structures of interest from the photographic data and computer graphic reconstructions. The image derived from digitizing is confirmed by histological staining and examining each section. Such reconstructions allow visualization in three dimensions, or in horizontal, coronal, or sagittal sections. These data allow for exact measurement of structures and their topological interrelationships. The auditory system depicted will include nervous pathways as well as inner and middle ear structures. A quantitatively exact reconstruction permits a precise determination of multiple physiological recording sites, for us a most important application. [Work supported by NIH.]

U19. Recovery from saturation in lateralization of high-frequency stimuli. Ervin R. Hafer and Thomas N. Buell (Department of Psychology, University of California, Berkeley, CA 94720)

Lateralization of trains of high-frequency clicks improves with both the number of clicks in a train (n) and interclick interval (ICI). Functions relating log threshold to log n are linear; however, the absolute values of the slopes decrease from the optimal -0.5 toward 0.0 with shorter ICIs (generally 10 ms or less). We have argued that shallower slopes are the result of a form-rate-dependent saturation which reduces the amount of binaural information derived from successive clicks in the train. The decay of the proposed saturation was measured here by presenting trains with an ICI of 2.5 ms but with each train broken by either one or three pauses. Of interest was the duration of the break needed to fully recover from saturation. A pause of 7.5 ms proved to be sufficient, implying that the saturation responsible for shallow slopes in log-threshold, log- n plots is the result of an active process rather than of fatigue. [Supported by NIH.]

Session V. Psychological Acoustics III: Psychophysical Tuning Curves and Loudness

Edward C. Carterette, Chairman

Department of Psychology, University of California, Los Angeles, California 90024

Chairman's Introduction—1:00

Contributed Papers

1:05

V1. A neural-counting model based on physiological characteristics of the peripheral auditory system: Application to loudness estimation and intensity discrimination. G. Lachs, R. Al-Shaikh, Q. Bi (Department of Electrical Engineering, The Pennsylvania State University, University Park, PA. 16802), and M. C. Teich (Department of Electrical Engineering, Columbia University, New York, NY 10027)

We have previously used an energy-based neural-counting model, incorporating spread of excitation, receptor saturation, spontaneous neural activity, and refractoriness in the primary auditory fibers, to predict the outcome of a number of neurophysiological and psychoacoustical experiments [M. C. Teich, *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 71, S18 (1982); G. Lachs *et al.*, *IEEE Trans. Syst., Man, Cybern.* SMC-13, No. 5 (1983)]. We wish also to incorporate the frequency characteristics of the middle-ear transmission function, and the cochlear mapping function. This places a number of constraints on the allowed parameters of the theory. We discuss the parameter values required for our theoretical system to effectively predict the outcome of loudness-estimation and intensity-discrimination experiments, both for pure-tone and variable-bandwidth noise stimuli. [Work supported by NSF.]

1:15

V2. Choice of stimulus parameters for measurement of auditory functions. Frederic L. Wightman, Israel Raz, Therese M. Velde, and John H. Grose (Auditory Research Laboratory (Audiology), Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

As a preliminary to a large-scale study of the interrelations among several auditory abilities, we conducted a separate experiment in order to determine which stimulus conditions would best discriminate among subjects. Using a forced-choice forward masking paradigm in all conditions, we obtained, from 12 normal listeners, eight 6-point psychophysical tuning curves with probe frequencies from 500 Hz to 4 kHz, two 10-point unmasking patterns (one each at 1 and 3 kHz), and two 4-point functions relating combination-tone ($2f_1-f_2$) level and f_2/f_1 ratio (one each at 1 and 3 kHz). Data from individual subjects were fit with low-order polynomials, and the coefficients of these polynomials were entered into an analysis of variance. The major conclusions which arose from this analysis are: (1) Frequency resolution, to the extent it is characterized by forward-masked PTCs, is adequately represented by two six-point PTCs, one at 1 and one at 3 kHz. (2) Two-tone unmasking is independent of frequency, and thus one unmasking pattern, at either 1 or 3 kHz can be used to represent unmasking. (3) Combination-tone generation is also independent of frequency, so the slope of only one CDT vs f_2/f_1 function, at either 1 or 3 kHz, can adequately characterize CDT generation. [Work supported by NINCDS.]

1:30

V3. Relations among auditory functions in normal listeners. Israel Raz, Frederic L. Wightman, John H. Grose, and Therese M. Velde (Auditory Research Laboratory (Audiology), Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

The interrelations among frequency selectivity, nonlinearities, and speech perception in noise were examined in 24 normal-hearing young

adults. Measures of psychophysical tuning curves, suppression (as indicated by the extent of unmasking caused by several suppressors), and the audibility region of $2f_1-f_2$ were obtained with an adaptive forward-masking detection paradigm. Speech perception in noise was evaluated with the nonsense syllable test via a computer-controlled ("best of three") adaptive procedure. The relations among functions were traced with multivariate analytic techniques. Based on these analyses, we conclude that: (1) Sharpness of tuning is intimately related to Combination Tone Generation (CTG) but not to suppression. (2) CTG is not related to suppression. (3) Aspects of speech perception in noise can be accounted for by information on frequency selectivity and nonlinearities. [Work supported by NINCDS.]

1:45

V4. Adaptive, forced-choice, loudness-balance tests in normal listeners. Craig C. Wier (Department of Speech Communication, University of Texas, Austin, TX 78712), Elizabeth Huckins (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195), and Robert S. Schlauch (Department of Speech Communication, University of Texas, Austin, TX 78712)

An adaptive, two-interval, forced-choice procedure (Jesteadt, 1980) was used to obtain ABLB and modified-MLB loudness-growth functions for a partially masked, 2000-Hz tone burst. Subjects were six normally hearing adults. Steinberg and Gardner (1937) and Stevens and Guirao (1967) showed that the comparison of the loudness of a tone in quiet with the loudness of the same tone partially masked produced "recruiting" loudness-growth functions from normally hearing listeners. For the ABLB condition in the present study, a pulsed, wideband masking noise was presented to only one ear. The level of the tone in noise was varied and compared with the level of the tone in quiet presented alternately to the opposite ear. In the modified-MLB procedure, the tone was presented monaurally, alternately in quiet and in the presence of a pulsed, wideband masking noise to the same ear. Two noise levels (3 and 33 dB N_0) and eight standard-tone levels were used. The loudness-growth functions obtained binaurally and monaurally were virtually identical. The results of these experiments validate Jesteadt's adaptive procedure for the measurement of loudness growth. A comparison of the ABLB and MLB data suggests that a similar, partial-masking procedure could be used to develop a monaural, single-frequency, loudness-balance test for clinical use. [Work supported by DRF and NINCDS.]

2:00

V5. The relation between loudness growth and intensity discrimination. Robert S. Schlauch, Craig C. Wier (Department of Speech Communication, University of Texas, Austin, TX 78712), and Elizabeth Huckins (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

The notion that the intensity-difference limen can be used as an indirect test of loudness recruitment is an old idea. However, to date, a firm quantitative relation between the two measures has not been established. We consider the problem as one in which loudness is related to the overall stimulus magnitude and intensity discrimination is a reflection of the accuracy with which a loudness judgment can be made. Since for pure tones loudness and intensity discrimination covary as functions of intensity, for

A specific intensity one quantity can be inferred from the other. To test this relation, adaptive ABLB loudness-growth functions were obtained for a 2000-Hz, 500-ms pure tone from a normally hearing listener (partially masked monaurally) and a listener with a moderate, unilateral sensorineural hearing loss. Intensity-discrimination performance was measured for the same 2000-Hz tone, at several levels, in both ears of each subject. Intensity-recruitment functions derived from these data are very similar to the obtained loudness-recruitment functions. [Work supported by DRF and NINCDS.]

2:15

V6. A comparison of the relative merits of four psychophysical procedures. Brian R. Shelton and Irene Scarrow (Department of Psychology, University of Western Ontario, London, Ontario, Canada N6A 5C2)

The efficiency of four psychophysical procedures to estimate the detection threshold of a brief 1000-Hz sinusoid presented in noise was assessed in naive observers. Six threshold estimates were obtained for ten subjects in 50-trial adaptive runs using each of four procedures: two-alternative adaptive staircase, two-alternative maximum-likelihood, three-alternative adaptive staircase and the three-alternative maximum-likelihood method. The four procedures produced equivalent thresholds, although the staircase procedures provided biased measurements in the initial runs. A trial-by-trial analysis of the variability of threshold estimates revealed a marked advantage to the use of three alternatives with the adaptive staircase, but not with the maximum-likelihood procedures. The data provide information regarding the relative efficiency of the four procedures, and recommend that particular procedures should be used in specific instances. In general, the commonly used two-alternative adaptive staircase is the least efficient procedure of the four, and the three-alternative adaptive staircase provides the most rapid and reliable threshold estimates. [Work supported by NSERC.]

2:30

V7. Frequency spread of TTS in humans and squirrel monkeys exposed to industrial noises. Donald W. Nielsen, Diane Brandt (Otolological

Research Laboratories, 7036 Education and Research Building, Henry Ford Hospital, Detroit, MI 48202), Donald N. Elliott, Peter Boisvert (Department of Psychology, Wayne State University, Detroit, MI 48201), and Ivan Hunter-Duvar (Hospital for Sick Children, Toronto, Ontario, Canada M5G 1X8)

We have been investigating the squirrel monkey (*Saimiri sciureus*) as a suitable animal model for testing the effects of noise on hearing. Our previous research has shown a similarity between human and squirrel monkey temporary threshold shift (TTS) growth with time in the noise. The goal of this study was to measure the frequency spread of TTS to five different industrial noises for both humans and squirrel monkeys and to compare the results. Fifteen human subjects and six squirrel monkeys were exposed to each of the five noises for 8 h, and TTS was measured at ten different frequencies ranging from 250 Hz to 8 kHz. In the experiments with humans, five frequencies were tested twice (counterbalanced) after each 8-h session. In the squirrel monkey experiments, a complete 8-h exposure was required for each frequency tested, since only one frequency could be measured each time the monkeys were removed from the noise. The results indicate that in predicting human TTS from the squirrel monkey TTS for these exposures, TTS would be overestimated at 750 Hz and 1 kHz and underestimated at 5.6 kHz, whereas the TTS at other frequencies would correspond closely. [Work supported by NIOSH.]

2:45

V8. Vibrotactile sensation as a function of hand preference. Ronald T. Verrillo (Institute for Sensory Research, Syracuse University, Syracuse, NY 13210)

A psychophysical matching procedure was used to assess the subjective magnitude of vibrotactile stimuli in right-, left-, and ambi-handed subjects. No threshold differences were noted. At suprathreshold levels of stimulation, enhancement of sensation by a preceding stimulus was the same in all groups when both stimuli were presented to either hand ipsilaterally. However, when stimulus pairs were presented contralaterally, only right- and left-handers showed enhancement and suppression effects. No contralateral effects were observed in ambi-handers. The results suggest that bilateral neural connections in right- and left-handers may be different from those of ambi-handers.

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

SENATE/COMMITTEE ROOMS, 1:00 P.M.

Session W. Underwater Acoustics III: Bottom Interaction (Précis-Poster Session)

DeWayne White, Chairman
NORDA, NSTL Station, Mississippi 39529

Chairman's Introduction—1:00

Contributed Papers

1:05

W1. The application of stepwise coupled modes to a refracting ocean using Galerkin's method. Richard B. Evans (Naval Ocean Research and Development Activity, NSTL Station, MS 39529)

The method of coupled modes developed [R. B. Evans, *J. Acoust. Soc. Am.* 74, 188-195 (1983)] for a homogeneous water column with stepwise depth variations of a penetrable attenuating bottom is applied to a realistic ocean environment with general sound-speed profiles. The main complication associated with this application is caused by having to find the complex eigenvalues for the range independent sections and not by the range dependence. To handle this problem, the Galerkin method is employed with an appropriately chosen orthonormal basis set which replaces

the search for the complex eigenvalues with a more tractable matrix eigenvalue problem. Numerical results based on this application are presented. [Work supported by NORDA.]

1:09

W2. A description of surface-wave scattering using a stepwise coupled-mode approach. C. Feuillade (ODSI Defence Systems Inc., 6110 Executive Boulevard, Rockville, MD 20852 and NORDA, NSTL Station, MS 39529)

We have extended the method of stepwise coupled modes to describe propagation through a medium of variable depth due to surface waves.

This facilitates the modeling of acoustic scattering from a rough surface. The method may be applied to describe scattering within a waveguide with a rough surface, or to model energy losses due to scattering out of a surface duct. Numerical procedures have been codified to solve the equations resulting from the analysis and some examples are provided.

1:13

W3. Sound field fluctuation in shallow water waveguide. C. S. Clay, Y. Y. Wang, and E. C. Shang (Department of Geology and Geophysics, University of Wisconsin-Madison, Madison, WI 53706)

The sound intensity fluctuation induced by the fluctuation of the surface in a shallow water waveguide was analyzed by using the adiabatic approximation of the normal mode theory. The sound field fluctuation can be categorized into two parts: (1) The phase fluctuation of the normal mode corresponds to the fluctuation of the horizontal component of the wavenumber (I. Tolstoy and C. S. Clay, *Ocean Acoustics*, pp. 222-236). (2) The fluctuation of the eigenfunction corresponds to the fluctuation of the vertical component of the wavenumber and the fluctuation of the local water depth and the local source and receiver positions. The randomness of the field intensity defined as $\eta = \text{SQR}[(\bar{I}^2 - \bar{I}^2)/\bar{I}^2]$ was calculated. It was found that the fluctuation of the second category can be the dominant comparing with the first category. The calculated result also agrees well with the experimental result conducted in a water tank [Y. Y. Wang *et al.*, *Chin. Phys.* 2, 515 (1982)].

1:17

W4. Sediment contributions to group velocity in shallow water. David W. Oakley and Robert A. Koch (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

Dispersion curves are used to examine the effect of sediments on group velocity, and the problem further dissected by a direct look at the relative contributions of the sediment layers to the group velocity integral. The environments of interest have water depths less than 150 m and sediment thickness of the order of 50 m. The range of frequencies considered is between 25 and 600 Hz. A systematic variation of sediment types and velocity profiles will be examined to find features of the dispersion curves which might be used to extract information about the bottom from shot data. In the presence of stable water velocity profiles, gradients in the sound velocity in the sediment can be detected, as can layering, provided such layers are near the water-sediment interface. Highly variable water sound velocity profiles limit information available concerning the sediment to gross features such as sediment type. [Work supported by Naval Ocean Research and Development Activity.]

1:21

W5. Shallow water waveguide characterization using Hankel transforms. James F. Lynch and George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A coherent processing technique is proposed for the characterization of shallow water channels. For a horizontally stratified ocean and bottom, the method consists of measuring the magnitude and phase versus range of the pressure due to a cw point source and Hankel transforming this data to obtain the depth-dependent Green's function versus horizontal wavenumber. The Green's function contains all the information about the channel necessary to solve the forward problem, including the nature of the discrete and continuous spectra and the plane-wave reflection coefficient of the bottom. Characteristics of the Green's function can also be used to infer acoustic properties of the bottom. Results for simple examples are presented using synthetic data and implications for the nonstratified case are discussed.

1:25

W6. Wave reflection from a sediment layer with depth-dependent properties. M. Stern, A. Bedford, and H. R. Millwater (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

The prediction of reflection coefficients for the water-sediment interface is strongly influenced by the physical nature of the sediment (a viscoelastic porous material saturated by a liquid) and the depth dependence of the mechanical properties [E. L. Hamilton, *J. Acoust. Soc. Am.* 68, 1313-1340 (1980)]. An efficient computational scheme for calculating reflection coefficients has been developed for a sediment layer using the Biot equations with depth-dependent coefficients. The qualitative effects of the inhomogeneous sediment properties (in comparison with uniform properties) are studied numerically. [Work supported by the Office of Naval Research.]

1:29

W7. Acoustic reflection from ocean abyssal plains. Herbert Freese (Arete' Associates, Arlington, VA 22202), Ioannis Besieris, and Werner Kohler (Virginia Tech, Blacksburg, VA 24061)

Existing bottom loss and pulse response data, measured at a variety of grazing angles, frequencies, and locations (Sohm and Hatteras abyssal plains, Blake Plateau, Florida shelf, and Gulf of Mexico), will be reviewed. A comparison of data with existing theoretical models will be made.

1:33

W8. An investigation of acoustic interaction with the ocean bottom from experimental time series generated by explosive sources. David Knobles (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

Received time series from explosive sources in a thin sediment environment are compared to simulated time series calculated by a ray theory model. The comparisons are made in various frequency bands to aid in the identification of sediment penetrating arrivals by taking advantage of the frequency dependence of the absorption of the sediment. For large ranges, most of the received energy refracts within the sediment. As the range decreases, the fraction of received energy due to reflections at the water-sediment interface increases. Discrepancies between the experimental and simulated time series will be presented and interpreted in terms of reflections from thin layers within the sediment and nonspecular basement reflections. [Work supported by Naval Ocean Research and Development Activity.]

1:37

W9. Estimated spatial coherence from a layered ocean bottom. T. W. Tunnell (NORDA Code 340, NSTL Station, MS 39529)

In this modeling study subsurface layering effects on the spatial coherence of bottom-interacting sound were investigated. The ocean bottom was modeled as alternating layers of clay and clayey silt. A silty clay bottom was used as a control. Complex reflection coefficients were calculated using a plane-wave multilayer model [G. J. Fryer, *J. Acoust. Soc. Am.* 63, 35-42 (1978)]. Using a frequency averaging technique, estimates of the spatial coherence were obtained as a function of grazing angle for fixed sensor separation. The results indicated that subsurface layering can seriously degrade the spatial coherence at frequencies above 100-200 Hz. The spatial coherence is considerably less affected by subsurface layering at very shallow grazing angles (10° or less) or very steep grazing angles (75° or greater). [Work supported by NORDA.]

1:41

W10. A WKB-numerical integration method for computing bottom loss. Terry Foreman and Richard Pitre (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

Computation of bottom loss at many frequencies and grazing angles is a computationally intensive process. At high frequencies and large grazing angles, the numerical integration of the wave equation can be very slow. When shear wave generation is an important process, the problem is exacerbated because low shear wave speeds imply short wavelengths and therefore small integration step sizes. The WKB approximation method is very rapid and accurate at short wavelengths, but is inaccurate near turning points. The hybrid integration method described here combines both

methods by dividing the integration domain into WKB and numerical integration (turning point) regions. Due to the frequency dependent size of the turning point region, the accuracy and computational effort of this hybrid method are only weakly dependent on frequency. [Work supported by Naval Ocean Research and Development Activity.]

1:45

W11. Optimum frequency of acoustic propagation in shallow water. D. A. Gershfeld and A. I. Eljer (Naval Research Laboratory, Washington, DC 20375)

The effect of various basic physical and environmental parameters on the optimum frequency of propagation in the ocean is investigated through the use of a model that is computationally simple to evaluate. A closed form expression for predicting transmission loss (for the Pekeris case) is obtained by applying the Poisson sum formula to transform a slowly converging transmission loss expression (consisting of a sum over the normal modes) into a rapidly converging series where only a few terms are needed to predict reasonable loss levels. A comparison of spectrum levels between the derived loss expression and a sophisticated normal mode transmission loss model shows close agreement for the frequencies of interest. The results show that the optimum frequency is strongly dependent upon the respective depths and separation distance of the source-receiver pair. The model is used to explore the dependence of the optimal frequency on several parameters. [Work supported by ONR Code 425UA and NAVSEA Code 63R.]

1:49

W12. Seafloor acoustic backscatter measurements using SEA BEAM. C. P. de Moustier (Marine Physical Laboratory of the Scripps Institution of Oceanography, University of California, San Diego, CA 92152)

Acoustic backscatter from the seafloor is measured using the 12-kHz SEA BEAM multibeam echo sounding system aboard the R/V THOMAS WASHINGTON. During a cruise in the eastern tropical Pacific, a very dense survey of a 15- x 20-mile manganese nodule area was carried out, and the acoustic returns from the 16 beams were recorded digitally on magnetic tape. Using the intensity of the specular return for each ping, a reflectivity map of the area was produced. The patchiness of the nodule coverage is evidenced by definite highs and lows in the reflectivity pattern. This map later served as the basis for a near bottom survey using the Deep Tow instrumentation package of the Marine Physical Laboratory and there is remarkable agreement between the two sets of measurements. Bottom photographs taken throughout the area also confirm the nodule distribution. This paper compares and discusses these results.

1:53

W13. Acoustic exploration of ocean floor properties based on the ringing of sediment layer resonances. Anton Nagl, Herbert Überall, and Kwang-Bock Yoo (Department of Physics, Catholic University, Washington, DC 20064)

It has recently been demonstrated [A. Nagl, H. Überall, and W. R. Hoover, IEEE Trans. Geosci. Electron. GE-20, 332 (1982)] that the physical properties of a layered ocean floor may in principle be determined from an analysis of the resonances in frequency, or in angle of incidence, which appear in bottom-reflected acoustic signals. Here, we study the possibility of probing the characteristics of isolated resonances by employing sinusoidal pulse trains of sufficient duration; this contrasts with conventional techniques based on the transmission times of short pulses, where many resonances contribute simultaneously to the reflected amplitude. It is found that the resonances manifest themselves by a ringing of the signal, and by a characteristic interference between the specular component and the fields resulting from multiple reflections within a layer. [Supported by the Office of Naval Research and by the College of Marine Studies, University of Delaware, where part of this work was carried out.]

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

SAN DIEGO ROOM, 1:00 TO 3:00 P.M.

Session X. Architectural Acoustics IV: Acoustics of Worship Spaces (Poster Session)

Ronald L. McKay, Co-Chairman

Bolt Beranek and Newman Inc., 21120 Vanowen Street, Canoga Park, California 91303

David Lubman, Co-Chairman

David Lubman and Associates, 2217 Vista Del Sol, Fullerton, California 92631

Chairman's Introduction—1:00

The acoustic environment of a worship space often must be hospitable to both speech and music, and this frequently must be accomplished while dealing with an architectural form that is not conducive to good hearing conditions. Considerable creativity is, therefore, required of the acoustical designer, usually employing both room-acoustic and electro-acoustic techniques. The creativity challenge seems to be increasing with an apparent trend toward worship spaces that can seat thousands and serve as performing arts facilities too.

Acoustical consultants and others who have been involved with the worship space design challenge have been invited to present posters displaying their solutions. These will form a basis for display, discussion, and archival record.

Posters will be presented by the following persons and organizations:

Ann Boyer, Jaffe Acoustics, Inc., Norwalk, Connecticut

Facilities: Chapel at University Park, Akron, Ohio; Serbian Orthodox Church, Aliquippa, Pennsylvania; United Presbyterian Church, Cuyahoga Falls, Ohio; Riverside Baptist Church, Denver, Colorado.

O. L. Angevine, Angevine Acoustical Consultants, Inc., West Falls, New York

Facilities: University of Rochester Interfaith Chapel, Rochester, New York; Lake Avenue Baptist Church, Rochester, New York; Randall Memorial Baptist Church, Amherst, New York.

Edward A. Daly, Daly Engineering Company, Beaverton, Oregon

Facilities: Saint Therese Catholic Church, Portland, Oregon; First United Methodist Church, Corvallis, Oregon; First Church of the Nazarene, Portland, Oregon; United Church of Christ Congregational, Forest Grove, Oregon.

David Joiner, Joiner-Pelton-Rose, Inc., Dallas, Texas

Facilities: First Baptist Church of Orlando, Orlando, Florida; Lovers Lane United Methodist Church, Dallas, Texas; Santuario de Guadalupe, Monterrey, Mexico; Grand Avenue Baptist Church, Fort Smith, Arkansas.

Bertram Y. Kinzey, Jr., Consultant in Architectural Acoustics, Gainesville, Florida

Facilities: First Lutheran Church, Gainesville, Florida; Crossroads Church of Christ, Gainesville, Florida.

Larry Kirkegaard, R. Lawrence Kirkegaard & Associates, Inc., Lombard, Illinois

Facilities: Calvary Lutheran Church, Golden Valley, Minnesota; House of Hope Presbyterian Church, Saint Paul, Minnesota; Trinity Episcopal Church, Portland, Oregon; Seventh Day Adventist Church, Collegedale, Tennessee.

Marshall Long, Marshall Long/Acoustics, Santa Monica, California

Facilities: St. Margaret Mary Church, Winter Park, Florida; Mission Valley Free Methodist Church, San Gabriel, California.

Daniel W. Martin, Daniel W. Martin Acoustical Consultant, Cincinnati, Ohio

Facilities: St. James Church of White Oak, Cincinnati, Ohio.

Ronald L. McKay, Bolt Beranek and Newman Inc., Canoga Park, California

Facilities: First Church of the Nazarene, Pasadena, California

Minoru Nagata, Minoru Nagata, Acoustic Engineer & Associates Co., Ltd., Tokyo, Japan

Facilities: St. Anselm's Priory, Tokyo, Japan; St. Ignatius Church, Tokyo, Japan; Shiga Sacred Garden, Shigaraki, Shiga, Japan.

Roger C. Noppe, Purcell + Noppe + Associates, Inc., Chatsworth, California

Facilities: Whitesburg Baptist Church, Huntsville, Alabama.

Dennis A. Paoletti, Paoletti/Lewitz/Associates Inc., San Francisco, California

Facilities: Hope Lutheran Church, Daly City, California; Community Presbyterian Church, Danville, California; Beth Eden Baptist Church, Oakland, California; Glad Tidings Pentecostal Church, Victoria, B. C., Canada.

Jack B. C. Purcell, Purcell + Noppe + Associates, Inc., Chatsworth, California

Facilities: St. Matthew's Episcopal Church, Pacific Palisades, California

L. W. Sepmeyer, Ludwig W. Sepmeyer, Consulting Engineer, Los Angeles, California

Facilities: Grace Bretheren Church, Long Beach, California; Grace Bretheren Chapel, Long Beach, California.

Stephen M. Sessler, Newcomb & Boyd, Atlanta, Georgia

Facilities: Temple Israel, Memphis, Tennessee.

Ewart A. Wetherill, Bolt Beranek and Newman Inc., Canoga Park, California

Facilities: Memorial Church, Stanford University, Palo Alto, California; Chapel, Occidental College, Los Angeles, California; St. Basil Catholic Church, Los Angeles, California.

Arthur K. Yeap, ADI Group, San Francisco, California

Facilities: Bethel Church, Lodi, California.

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

COUNCIL ROOM, 1:00 TO 2:50 P.M.

Session Y. Noise III: Community Noise

Gerald J. Franz, Chairman

Silencing Technology Associates, P. O. Box 695, Bayview, Idaho 83803

Chairman's Introduction—1:00

Contributed Papers

1:05

Y1. Highway identification of exhaust system noise problems. F. M. Kessler (Dames & Moore, 6 Commerce Drive, Cranford, NJ 07016), E. DiPolvere (NJDEP, Trenton, NJ 08618), and M. Van Ouwerkerk (Dames & Moore, Cranford, NJ 07016)

To reduce the number of excessively noisy vehicles on the road, a simplified noise control enforcement procedure was developed and tested. This procedure utilizes the "trained" ear of enforcement officials in detecting faulty, modified, or absent mufflers and exhaust systems. The purpose of this study was to evaluate whether or not the trained ear can be utilized as a screening device for "probable cause" to stop a vehicle for inspection of the exhaust system resulting in the citation of the driver for a mechanically defective exhaust system. Two teams conducted these tests. The first team included the "spotting officer" and a person who recorded

the pass-by sound level of all vehicles. The second team consisted of an inspection officer and a person making stationary sound level measurements. A majority of the vehicles called loud by the "spotting" officer had pass-by sound levels 5-10 dB below the pass-by standard for that class of vehicle. This indicates that the quality of the sound emitted by such vehicles is distinctive and that the officers are able to detect defective exhausts prior to the defects causing the excessive noise to exceed the pass-by standard. Numerous additional findings are discussed.

1:20

Y2. Some similarities in community response to aircraft and road traffic noise. S. Fidell (Bolt Beranek and Newman Inc., 21120 Vanowen Street, Canoga Park, CA 91305)

A social survey was undertaken in the vicinity of a major air carrier airport. One of the two neighborhoods was exposed to noise due to aircraft operations at about $L_{dn} = 70$ dB. The other neighborhood was exposed to road traffic noise at about $L_{dn} = 62$ dB. A brief structured interview conducted by telephone revealed a number of similarities in community response to the two noise exposure environments. These included reports of sleep interference and habituation to noise exposure. Differences in observed prevalence of annoyance in the two neighborhoods were attributable to the differences in exposure levels.

1:35

Y3. The effects of community noise on modular and manufactured/mobile homes. Mones E. Hawley and Geoffrey M. Back (Jack Faucett Associates, 5454 Wisconsin Avenue, Chevy Chase, MD 2815)

Dwellings which are built in factories and transported to their sites account for an increasingly important fraction of new housing construction in the United States. About half of these houses are placed in clusters with other factory-built homes. The interior sources of noise and the insulation against noise from exterior sources were investigated and compared with those for new site-built houses. Manufactured houses (formerly called mobile homes), which have an integral chassis for transportation, must be built in conformance with federal standards. Modular homes, which are removable from their transporters, are built to state and local building codes. The result is that factory-built houses have internal noise levels and external-to-internal noise level reductions equivalent to those of comparable site-built houses. However, exterior community noise levels frequently are higher for factory-built homes than for site-built homes because of the way the sites are chosen. [Work supported in part by the U.S. Environmental Protection Agency under contract number 68-01-6159.]

1:50

Y4. The effect of curved vehicle path on average sound pressure level. Gregory S. Davis and C. E. Wallace (Department of Mechanical and Aerospace Engineering, Arizona State University, Tempe, AZ 85287)

A mathematical theory is presented which predicts average noise levels for curved aircraft flight paths. The error in computing equivalent noise levels by assuming a straight flight path and then bending it to follow the curved flight path is evaluated as a function of the shape of the curved flight path and the location of the observer. The theory is experimentally verified using an automobile on a curved test track. It is shown that for a given observer location, the amount of error increases as the amount of path curvature increases. Also it is shown that for a given curved path, the amount of error increases as the observer location is moved away from the curved path. In some cases the error incurred is significant and is in excess of 3 dB.

2:05

Y5. Airport related residential acoustical insulation demonstration project. J. T. Weissenburger (Engineering Dynamics International, St. Louis, MO 63124)

Faced with making a choice between acquiring residences and clearing the property versus acoustically insulating them, the St. Louis Airport Authority undertook a demonstration project to develop firm data on the cost and value of retrofit acoustical insulations for residences in the airport environs. Six wood frame homes, typical of most of the housing stock in the area, were modified using standard available construction techniques and materials in an effort to improve their insulation against aircraft noise. Noise exposure measurements were made before and after modifications. This paper reports the details of the retrofit modifications employed and their effectiveness in reducing exposure to aircraft noise.

2:20

Y6. An overground excess sound attenuation (ESA) study. Robert G. Powell (AFAMRL/BBE, Wright-Patterson Air Force Base, OH 45433)

In order to better predict community noise from Air Force operations an extensive study on overground excess sound attenuation (all attenuation in excess of inverse square and air absorption losses) was conducted at Wright-Patterson Air Force Base. The unique aspects of this study were the methods used to acquire the reference noise levels, procedures used to obtain up and downwind propagated noise, and the all-weather, radio-controlled microphone recording systems. To be presented are the experimental design and samples of ESA values calculated from the noise of 400 plus ground runups of a KC-135A aircraft propagating along a 2400-m microphone array. Also to be discussed are the availability and format of the basic acoustic and meteorological data from this study.

2:35

Y7. Analysis of overground excess attenuation. Dwight E. Bishop (Bolt Beranek and Newman Inc., 21120 Vanowen Street, Canoga Park, CA 91303)

The noise and meteorological data acquired by the Air Force AMRL at Wright-Patterson Air Force Base has been analyzed to determine the factors most influencing the excess sound attenuation at different frequencies and distances. Multiple regression analyses of the excess sound attenuation (the difference in noise levels measured at different positions referred to the reference position levels after adjustment for inverse square and air absorption losses) indicate that the strongest variables influencing the excess sound attenuation are the wind component in the direction of sound propagation, the temperature gradient, and the ground cover (grass versus snow). Regression coefficients ranged up to the order of $r^2 = 0.8$. Plots of the variance accounted for (r^2) values as a function of frequency show two distinct peaks which shift downwards in frequency as distance increases. At low frequencies (below about 200 Hz), the most significant variable is ground cover. For the higher frequencies the most significant variable is the wind component. Curves showing the measured excess attenuation values as a function of wind component and temperature gradient will be compared to results from other studies and theoretical models.

Session Z. Physical Acoustics III: General Topics II

Robert T. Beyer, Chairman

Department of Physics, Brown University, Providence, Rhode Island 02912

Chairman's Introduction—1:00

Contributed Papers

1:05

Z1. Removal of the assumption of plane primary waves in the theory of the nonscattering of sound by sound. Peter J. Westervelt (Applied Research Laboratories of the University of Texas, P.O. Box 8029, Austin TX 78712)

The virtual source of combination frequencies is well known to be $Q = \beta (\partial/\partial t)(p_1 p_2)$, where $\beta = \rho_0^{-2} c_0^{-4} [2 + (d^2 p/d\rho^2)_{\rho_0}]$ and the pressure of each of the two primary waves may be expressed as $p_1 = a_1 \cos \psi_1$, and $p_2 = a_2 \cos \psi_2$, where the a 's and the eikonals, the ψ 's, depend on the four-radius vector x^i (refer to "eikonal" in the index of any late edition of Landau and Lifshitz, *The Classical Theory of Fields*). The restriction to plane primary waves implicit in my derivation of Q [J. Acoust. Soc. Am. 35, 535-537 (1963)] was later removed by me [Proc. Univ. of Texas Nonlinear Acoustics Symposium, edited by T. G. Muir, pp. 167-181 (1970)]. The present paper reports a function R whose d'Alembertian equals Q when the region of interaction common to both primary waves resides many primary wavelengths from each of the two sources. It turns out that $R = \beta (\partial/\partial t) [a_1 a_2 [(\partial\psi_1/\partial x_i)(\partial\psi_2/\partial x^i)]^{-1} \sin \psi_1 \sin \psi_2]$ in which $(\partial\psi_1/\partial x_i)(\partial\psi_2/\partial x^i)$ means $\nabla\psi_1 \cdot \nabla\psi_2 - c_0^{-2} \psi_1 \psi_2$. Thus $\nabla^2 R - c_0^{-2} \ddot{R} = Q$ plus terms that are negligible if the primaries are not collinear since these terms are of order E^{-1} and E^{-2} smaller than Q where $E = \psi + i\omega t$ is the space-only dependent eikonal. E_1 and E_2 have magnitudes of the order $r_1 \lambda_1^{-1}$ and $r_2 \lambda_2^{-1}$, respectively, where r_1 and r_2 are the distances from the interaction region to the primary sources of waves having dominant wavelengths λ_1 and λ_2 , respectively. The conclusion is that no scattering occurs outside the interaction region of two waves, neither one of which needs to be strictly plane, nor for that matter steady state.

1:17

Z2. Acoustic signals having both positive and negative nonlinearity. M. S. Cramer (Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061) and A. Kluwick (Institut fuer Stroemungslehre und Waermuebertragung, Technische Universitaet Wien, A-1040, Vienna, Austria)

The quantity $\Gamma \equiv (1/\rho)(\partial\rho a/\partial\rho)$ frequently occurs in the nonlinear acoustics of both gases and liquids. Here ρ is the density, a the sound speed, and the derivative is taken at constant entropy. It is well known that when $\Gamma > 0$ (positive nonlinearity), pulses and periodic wave trains steepen forward to form compression shocks. When $\Gamma < 0$ (negative nonlinearity), the steepening is backward and expansion shocks form. When the signal amplitude is sufficiently large compared to Γ , the local value of Γ may vary from point to point in the wave and the subsequent behavior is quite different than the cases having either positive or negative nonlinearity everywhere. This phenomena can occur for single phase Navier-Stokes fluids and the present paper will provide a description of the propagation in this latter case. Results of interest include shock splitting, shock waves having sonic conditions either upstream or downstream of the shock, and collisions between compression and expansion shocks. Under certain conditions it was found that the ultimate decay of a mono-signed pulse is proportional to the negative one-third power of the propagation time rather than the classical negative one-half power.

1:29

Z3. Sound generation by a moving laser source. Yves H. Berthelot and Nicholas P. Chotiros (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78712)

A theoretical and experimental study of moving thermoacoustic sources is presented in this paper. The inhomogeneous linear wave equation for a fluid containing heat sources [P. J. Westervelt and R. S. Larson, J. Acoust. Soc. Am. 54, 121-122 (1973)] is used to derive the acoustic response $h(t)$ of a medium excited by an impulse of light. The pressure radiated by a laser source moving at constant speed over a water surface is then described in terms of a convolution between the impulse response $h(t)$ and the optoacoustic source strength which is proportional to dI/dt , where $I(t)$ represents the laser intensity. In some limiting cases of farfield radiation, the computed pressure waveforms can be compared with previous work [L. M. Lyamshev and L. V. Sedov, Sov. Phys. Acoust. 27(1), 4-18 (1981)]. An experimental study is also in progress. The motion of the laser beam over the water surface at velocities up to Mach 2 is achieved by means of a rotating mirror. The laser is excited in the conventional non-Q-switched mode, with a pulse duration of 1 ms. [Work supported by ONR.]

1:41

Z4. Dark-field magneto-elastic nondestructive testing technique. Jacques R. Chamuel¹⁾ (The Charles Stark Draper Laboratory, Inc., 555 Technology Square, Cambridge, MA 02139)

In the field of nondestructive testing, great advantages can be obtained if a dark-field technique can be achieved. A magneto-elastic technique generating sound waves only in the presence of a defect and from the defect is described (no defect—no sound waves in the material under test). The magneto-elastic technique generates three-dimensional information about the defect location, dimensions, and characteristics, unlike conventional ultrasonic NDE techniques where the defect is used as a scatterer of incident ultrasonic waves. Eddy currents are generated in the material under test with a current carrying conductor. A magnet is placed such that its magnetic field is parallel to the current carrying conductor. Elastic waves are generated in the conductive material under test only at the crack edges from the interaction of the magnet with transverse eddy currents flowing around the defect. The generated elastic waves are detected with an ultrasonic receiver transducer coupled to the material. An array of receivers may be used to triangulate on the defect and also to characterize the properties of the defect from the 3-D information. Conventional EMAT (electromagnetic acoustic transducers) are used to generate elastic waves in the material even when no defects are present; the magneto-elastic technique depends on the presence of a defect to create the elastic waves. Preliminary experimental results are presented together with other potential applications of the new concept.²⁾ The author is currently with Sonoquest, P.O. Box 584, Sudbury, MA 01776.

1:53

Z5. Self-scanning-line electro-elastic NDE technique. Jacques R. Chamuel¹⁾ (The Charles Stark Draper Laboratory, Inc., 555 Technology Square, Cambridge, MA 02139)

A contactless self-scanning-line electro-elastic nondestructive testing technique is described capable of determining in real time the presence, location, number, and size of defects present along a linear path on the surface of a conductive material under test. In one embodiment, the NDE probe consists of a magnetostrictive delay line placed in close proximity with the surface to be inspected. A current carrying wire placed adjacent and parallel to the delay line is used to induce eddy currents in the material. The presence of a crack creates transverse eddy currents interacting

locally with the magnetostrictive delay line launching elastic waves in the delay line. The elastic waves are generated only at the crack edge locations along the delay line. Spatial information regarding crack edge locations is preserved in the time information. An ultrasonic receiver coupled to the delay line is used to detect the elastic waves traveling in the delay line. The self-scanning-line electro-elastic NDE probe is not susceptible to the presence of elastic waves propagating in the material under test. The new concept is applicable to flat and curved surfaces including pipe inspection. Conventional eddy current techniques are not self scanning along a linear path giving only an integrated response, indicating only the presence and not the location and number of defects. Application of the concept to automation will be discussed.⁴⁾ The author is currently with Sonoquest, P.O. Box 584, Sudbury, MA 01776.

2:05

Z6. Nonlinear transient response of electrostatic loudspeakers. Juergen Schroeter (Bell Laboratories, Murray Hill, NJ 07974 and Lehrstuhl f. Allg. E-Technik und Akustik, Ruhr-Universitaet, Postfach 102148, D-4630 Bochum, Federal Republic of Germany) and Man Mohan Sondhi (Bell Laboratories, Murray Hill, NJ 07974)

When an electrostatic loudspeaker (ESL) is used as an acoustic pulse generator [see M. M. Sondhi and J. R. Resnick, *J. Acoust. Soc. Am.* **73**, 985-1002 (1983) for one such recent application], it may well be desirable to operate it at levels at which its motion is highly nonlinear. Available analyses of ESLs are inadequate for such applications because they are concerned with computing sine-wave response and estimating low-level harmonic distortion. To fill this gap we have developed an algorithm to compute the general, nonlinear transient response of a circular ESL to a circularly symmetric excitation. Our formulation is general enough to allow multiple support of the membrane at an arbitrary number of concentric circular ridges on the backplate. Our analysis includes the effects of a back cavity, a perforated backplate, and the coupling of the membrane motion to the eigenmodes of a cylindrical impedance tube in front of the membrane. (The program can be easily modified for other configurations, e.g., when the membrane is in an infinite baffle.) We will present some computed results and their comparisons to measured data.

2:17

Z7. Analysis of acoustically excited structures using holographic interferometry with thermoplastic films. Joseph A. Clark (Acousto-Optics Laboratory, Mechanical Engineering Department, Catholic University of America, Washington, DC 20064)

Experimental studies of the transient and modal response of vibrating bodies excited by sound fields will be described. Reusable thermoplastic film is utilized with an automated hologram recorder in order to more conveniently obtain real-time and time-average holograms. This approach facilitates interactive analysis and design of vibrating structures. Typical structures being currently studied are membranes excited both resonantly and nonresonantly by airborne sound. Membrane structures appropriate for spatially resolved measurements of radiating sound fields and turbulent boundary pressure fields will be discussed. [Research sponsored by O.N.R. and D.T.N.S.R.D.C.]

2:29

Z8. Experimental study of internal noise on a shell submitted to boundary layer excitation. Jacqueline Larcher, Jean-Marc Parot, and Jean-Paul Berhault (Société Metravib, BP 182, 69132 Ecully Cedex 2, France)

A study has been done of the acoustic performance of elastic material shells, surrounded by a heavy fluid, submitted to a turbulent boundary layer excitation. Measurements of internal noise levels under localized excitation, in frequency bands, show good agreement with the result given by a statistical energy model. These results show significant effects of the structural properties and material for a given shell shape. The results agree well with noise and vibration measurements in a reciprocity situation (action of an internal noise source on the structure). Flexural waves in the shell material, although subsonic, play a major role.

2:41

Z9. Experimental investigation of nonlinear crossed beam scattering in the presence of turbulence. Murray S. Korman (Department of Physics, U.S. Naval Academy, Annapolis, MD 21402)

An experimental apparatus has been constructed to measure the nonlinear scattering of two mutually perpendicular crossed sound beams in the presence of turbulence (from a submerged water jet). Results are used to measure and characterize the turbulent flow. The pulsed beams (1.90 and 2.10 MHz) are generated from 1-in.-diam transducers, suspended from perpendicular radius arms that are free to rotate 360° in a horizontal plane. The transmitters and 4-MHz receiver are located 1 and 2 m from the interaction region, respectively. One experimental difficulty is the inability to distinguish between scattering from the interaction region and other nonlinear processes. As part of the calibration procedure the nonlinear scattering of a cylinder is investigated. Measurements show that scattering at angles near the main primary lobes are extremely difficult to measure due to the inability to separate out the scattered field of (a) one primary beam interacting with (b) the scattered linear field of the other primary beam from the cylinder. Preliminary results from the scattering by turbulence and by a cylinder will be presented. [Work supported by NRL (Physical Acoustics Branch) and USNA.]

2:53

Z10. Sound radiation from a concave or a convex dome in a semi-infinite tube. Hideo Suzuki (CBS Technology Center, 227 High Ridge Road, Stamford, CT 06905)

A previous study [H. Suzuki and J. Tichy, *J. Acoust. Soc. Am.* **69**, 41-49 (1981)] showed that a concave or a convex dome in an infinite baffle has radiation characteristics that are different from those of the flat piston. These results indicated that a concave or a convex dome in a lossless tube may also have different radiation characteristics than a similarly mounted flat piston. This problem was investigated using the same method as the previous studies. The results show that the farfield sound pressure response rises by about 0.5 to 0.7 dB in the frequency range $ka = 2.2$ to 2.8 for both concave and convex domes having the height-to-radius (a) ratio of 0.5 to 1.0. As expected, the radiation resistance increases by about 13% to 17% over the radiation resistance in the low-frequency region. The radiation reactance is very small at low frequencies but becomes comparable to the resistance at frequencies above $ka = 2.5$. The reactance is always negative below the cutoff frequency.

Session AA. Engineering Acoustics III: Transducer Materials

Robert M. Gorman, Chairman

Bolt Beranek and Newman, Inc., Union Station, New London, Connecticut 06426

Chairman's Introduction—1:30

Contributed Papers

1:35

AA1. Shock resistant PVDF hydrophones. T. A. Henriquez and Mary Lou Miller (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856)

Previous research successfully demonstrated the use of tubular PVDF in a shock resistant hydrophone [T. A. Henriquez, "Application of Tubular PVDF to Shock Resistant Hydrophones," in *Proceedings of the International Symposium of Applied Ferroelectrics* (1983)]. The present work continued the development and refinement of a shock hardened underwater transducer. Nonvoided as well as voided tubular material with an outer diameter of $\frac{1}{4}$ or $\frac{1}{2}$ in. was used as the active element in each design. To evaluate the effects of stress rod material and circumferential prestress, the free-field voltage sensitivity (FFVS) was recorded for each transducer. The response of the element was dependent upon the acoustic coupling between the compression bolt and rubber mandril within the tube and between the rubber mandril and the tube itself. The hydrophones were also exposed to scaled down explosive tests to determine their ruggedness. The results support previous findings that tubular PVDF can be utilized as a broadband standard hydrophone. [Work supported by NAVSEA.]

1:50

AA2. Underwater transducer wetting agents. L. E. Ivey and C. M. Thompson (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856)

The problem of air bubbles forming and clinging to the surface of oily or dirty underwater acoustic transducers has been around since the first calibration facilities were established in 1941. An air bubble driven at its resonance frequency has a measurable interference effect on the transducer's calibration response. The bubble problem can be eliminated by thoroughly treating the transducer with a wetting agent after the grease and oil have been cleaned off with a strong detergent. The wetting agent in some cases can also act as the degreaser. There is a controversy among present-day calibration facilities as to which wetting agent is best. A survey was made of the Navy-sponsored underwater transducer calibration facilities to determine which wetting agent each was using. Each was using a different type. Samples of each type were obtained and evaluated at NRL-USRD. The tests performed were surface tension measurements, compatibility measurements with several popular types of rubber materials, and acoustic tests of each type on standard transducers in the USRD's Lake Calibration Facility. The results of these tests shown in the form of curves, equations, and tabulated data validate which wetting agents are best to use as surface acting agents.

2:05

AA3. Corona characteristics of transducer ceramics. R. W. Timme, L. P. Browder (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856), and R. E. Montgomery (Texas Research Institute, Inc.)

Transducers are frequently plagued by the presence of corona which can adversely affect the performance and lifetime of the unit. Sites at which corona can occur include the wiring, spacing, configuration, sharp electrode edges, etc., but these can be minimized with proper design and

engineering. A fundamental limit below which one cannot reduce the appearance of corona lies with the transduction material, that is, the piezoelectric ceramic itself. The corona characteristics of over 2500 lead zirconate-lead titanate ceramic rings have been determined and the results used to model a much larger ring population from which predictions can be made about the corona performance of transducers assembled from these rings. A summary of the information to be presented is that: (1) Populations of "as-fired" ceramic rings have significantly higher corona inception voltages than rings with surfaces finished by machine grinding. (2) Two distinctively different types of corona have been identified, one of which can be partially "cured" by sulfur hexafluoride. (3) The corona data for the rings have been fit to a Weibull distribution model and applied to a "weak-link" model to predict the best performance that can be expected from the assembled transducer. [Work sponsored by the Sonar Transducer Reliability Improvement Program, NAVSEA.]

2:20

AA4. Analysis of segmented cylinder ceramic transducers using multiport interconnection techniques. Michael P. Johnson and Stephen C. Thompson (Systems Engineering Department, Ocean Systems Division, Gould Defense Systems, Inc., Euclid Avenue, Cleveland, OH 44117)

For many years transducer analysis using multiport interconnection techniques has been successfully applied to tonpilz designs and other transducer types in which the vibratory motion is entirely one-dimensional. The analysis of radially vibrating cylindrical ceramic elements has generally used either empirically derived equivalent circuit methods or has started directly from the differential equations of motion. For segmented cylinders it is possible to perform an approximate analysis using linear network analysis methods. This allows the use of standard multiport analysis computer codes such as SEADUCER to perform much of the calculation. Furthermore these methods eliminate the need for empiricism. Calculated radiation impedance and finite mounting surface impedance effects can be directly included. A description of the method and comparison of the results with previous methods will be provided.

2:35

AA5. On the construction of linear-response hydrophones. Mark B. Moffett (Naval Underwater Systems Center, New London, CT 06320), Theodore A. Henriquez, and Mary Lou Miller (Naval Research Laboratory, Underwater Sound Reference Detachment, Orlando, FL 32856)

Most hydrophone elements in use nowadays can be classified either as volume-mode elements or as piezoceramic elements. Although volume-mode elements are usually quite linear, nearly all have low capacitance, and so, in practice, are used in conjunction with preamplifiers. Preamplifiers, however, can be nonlinear and are not usually suitable for close-range parametric source measurements. Piezoceramic elements have high capacitances and therefore can be used without preamplifiers. However, they do not work well in the volume mode, and so they are usually configured so as to expose only one element surface to the acoustic pressure. Such configurations invariably require bonds of some sort, e.g., two ceramic hemispheres are glued together to form a spherical hydrophone.

Nonlinearity of an epoxy bond appears to account for the nonlinearity of F42D spherical hydrophones. Linear-response hydrophones suitable for use with parametric sources at close range can be constructed with volume-mode elements. If cable lengths can be kept short (< 10 m), no preamplifier should be used. Otherwise a passive low-pass filter should precede the preamplifier. [Work supported by Naval Material Command and Naval Sea Systems command.]

2:50

AA6. EPDM rubber as an underwater acoustic window. Corley M. Thompson, Rodger Capps, and Michael J. Lizzi (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856)

The ethylene-propylene-terpolymer (EPDM) class of elastomers has been widely suggested but rarely used in underwater acoustic devices. Recommendations for its use usually result from a recognition of its low acoustic losses and good match of sound speed and density with seawater. The hesitations about using EPDM's result from unknowns in its engineering properties of bondability and tear strength. The reported marginal bondability and tear strength were investigated as a function of EPDM formulation. Specifically, the effects of filler type and amount, sulfur or peroxide cure type and loading, and polymer parameters on both engineering and acoustic properties will be shown. With care in designing the EPDM formulation, this material can have acoustic properties (both rho-c and attenuation) as good as natural rubber and environmental resistance superior to neoprenes. EPDM's compatibility with castor oil is the best of all common classes of elastomers. The results of many tests confirm that while EPDM's bondability and tear strength are not equal to neoprenes, they are nevertheless adequate for most applications.

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

DEANZA/MESA ROOM, 1:00 TO 3:05 P.M.

Session BB. Speech Communication III: Voice Characteristics

Kung Pu Li, Chairman

ITT—DCD, 10060 Carroll Canyon, San Diego, California 92131

Chairman's Introduction—1:00

Contributed Papers

1:05

BB1. Recognition of famous voices forwards and backwards. Diana Van Lancker (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024), Jody Kreiman (Department of Linguistics, University of Chicago, Chicago, IL 60637 and Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024), and Karen Emmorey (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024)

To investigate familiar voice recognition, samples of famous male voices were tape-recorded and edited on a PDP-11/34 computer. Three listening tasks were prepared. Subjects indicated whether they recognized voices from (1) 2-s samples; (2) different and reordered 2-s samples presented along with six choices; and (3) 4-s samples presented backwards (rerandomized and refoiled). Ninety-six subjects were divided into four groups, three by age and one given the backwards presentation first. For task 1, the mean recognition rate was 17%. In task 2, subjects correctly identified 69.5% of the voices they knew (ascertained by questionnaires). Recognition for known voices presented backwards was 12.8% less than for voices presented forwards. In the group given the voices backwards first, a similar difference (12.5%) was observed. A two-way ANOVA comparing the four groups on tasks 2 and 3 revealed main effects of group and task, but no group-by-task interaction; thus differences in performance between forwards and backwards presentations were unchanged across the four subject groups. These results indicate that voice recognition can be achieved given only limited information which includes rate, pitch, and pitch range. [Work supported by NIH Grant No. 4-443944-31117.]

1:17

BB2. Recognition of famous voices given excerpted vowels, words, and 2-s texts. Karen Emmorey, Diana Van Lancker (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024), and Jody Kreiman (Department of Linguistics, University of Chicago, Chicago, IL 60637 and Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024)

The ability of 40 subjects to recognize 25 famous male voices was investigated. Samples were tape-recorded and edited on a PDP-11/34 computer. Three different listening tasks were prepared. Subjects indicated whether they recognized voices given six choices: (1) from 2-s samples; (2) from excerpted words (mean duration 481 ms; and (3) from excerpted and concatenated strings of vowels (mean duration 494 ms). The same voices were used in all tasks. Subjects were divided into two groups. For one group, the tasks were presented in the order 2-s texts, words, vowels. For group two, the reverse order was presented. A two-way ANOVA revealed no effect of presentation order. Subjects correctly identified 61% of the voices they knew (ascertained by a questionnaire) given 2-s texts, 40% of the voices given excerpted words, and 34% given vowel strings. Subjects performed significantly better on the 2-s task than on the word or vowels tasks. There was no significant difference between their performance on the word and vowels conditions. These results show that voices can be recognized given limited amounts of voice information. [Work supported by NIH Grant No. 4-443944-31117.]

1:29

BB3. A speaker recognizability test for communications systems. Panos Papamichalis and George Doddington (Texas Instruments Inc., P.O. Box 226015, MS 238, Dallas, TX 75266)

A Speaker Recognizability Test (SRT) has been designed which tries to establish how well a given communications system preserves a speaker's identity. Contrary to previous efforts, no attempt is made to identify the cues used by listeners for speaker recognition. Instead, listeners are asked directly to identify a speaker who says an utterance. The test is constructed as follows: Several sentences are collected from five male and five female speakers. One sentence from each speaker is used as reference. The listening team consists of ten listeners. Each listener is presented 20 different sentences and is asked to identify the speaker of each one of them by comparing it with the ten reference sentences. Among the issues considered in the design of the test is the choice of speakers, the use of reference sentences from the same or different sessions of data collection, and the use of processed or unprocessed speech for reference.

BB4. Intonation, pitch height, and voice quality as indicators of speaker attitude. Klaus R. Scherer, Kim Silverman, and D. Robert Ladd (Department of Psychology, University of Giessen, 6300 Giessen, West Germany)

Utterances from recorded interviews were rated for attitude conveyed. Stimuli were presented in both the original audio version and various degraded versions. Judgments of backwards and random-spliced versions correlated highly with originals, while judgments of low-pass filtered pitch contours did not. This suggests attitude is conveyed directly by voice quality largely independent of pitch contour. However, judgments of originals show (1) correlations between absolute pitch height and ratings of speaker arousal, and (2) interactions between linguistic categories of intonation and syntax in judgments of more "cognitive" attitudes like "friendly" and "critical" (e.g., final rises were rated friendly on yes/no questions but critical in WH questions). This shows that it is important to consider the linguistic structure of intonation and the accompanying text in studying intonational cues to attitude. Masking or merely averaging acoustic characteristics of F_0 will obscure these interactions. [Work supported by German Science Foundation.]

1:53

BB5. Suppression of noise in speech using PARCOR analysis. Atsushi Mano and Shinji Ozawa (Ozawa Laboratory, Department of Electric Engineering, Faculty of Science and Technology, Keio University, 3-14-1, Hiyoshi, Kouhoku-ku, Yokohama, 223 Japan)

In this paper, a new noise suppression algorithm is presented for reducing any kind of background noise in speech. This noise suppressor uses three kinds of filters. The first filter, which is the inverse lattice filter implemented by PARCOR analysis of noise part, makes any colored background noises into white noises. The second filter, which is the white noise suppressor using spectral subtraction, reduces the white noise components in speech. The third filter, which is the noninverse type of the first one, restores the speech distorted by the first filter. The algorithm which detects nonspeech periods is combined with this noise suppressor, and the characteristics of these filters adaptively follow the nonstationary noise environment. The traffic noise and jet engine noise, etc., were used to estimate this system. As a result of the tests, we could improve the speech to noise ratio more than 20 dB. And at the same time, we could improve the articulation of the speech.

2:05

BB6. A model for the synthesis of natural sounding vowels. Donald R. Allen and William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo UT 84602)

A model has been developed which is designed to preserve some of the naturalness that is usually lost in speech synthesis. A parameterized function is used to produce an approximation to the cross-sectional area through the glottis. A circuit model of the subglottal and glottal system is used to generate the volume velocity of the air through the glottis from the lung pressure and the time-varying supraglottal pressure. The tract is represented by its input impedance impulse response which can be calculated from the area function of the tract. A convolution of the input impedance impulse response with the volume velocity determines the supraglottal pressure. The equations relating the above two conditions for the volume velocity are solved simultaneously. The output of the model is generated by convolving the resulting glottal volume velocity with the transfer function impulse response of the tract. A comparison is made between vowels synthesized with and without the vocal tract glottal flow interaction. Listening tests showed that those vowels with the interaction were preferred as more natural sounding over those without the interaction.

2:17

BB7. Pitch encoding using simple unit pitch patterns. Kathleen M. Goudie, Kun-Shan Lin, and Gene A. Frantz (Consumer Products Group, Texas Instruments, P.O. Box 10508, M.S. 5707, Lubbock, TX 79408)

In encoding synthetic speech for vocabulary storage and reproduction, current techniques require storing at least five or six bits of coded pitch value per frame. Due to the slow variation of pitch contours, it is possible to describe these contours as a sequence of simple pitch patterns. If each pattern in the sequence is applied to a subunit of the coded phrase or word, each subcontour can furthermore be approximated in terms of simple linear functions. The pitch pattern is selected by picking the best match between the target or natural subcontour and one contour from a library of predefined simple pitch contours. By storing one pitch pattern contour per specific subunit in speech, a considerable savings in vocabulary storage may be achieved. Informal listening tests show that the degradation in speech quality is minor.

2:29

BB8. Periodic repetition of multi-pulse excitation. Bishnu S. Atal and Barbara E. Caspers (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

B. S. Atal and J. R. Remde (Proc. ICASSP 82) described a method for determining the multipulse excitation for LPC speech synthesis. We find that, even for reasonably periodic speech waveforms, the secondary pulses in the multipulse excitation do not show similar periodic behavior. What is the influence of this lack of periodicity in the excitation on the quality of synthetic speech? We have investigated this question by introducing controlled periodicity into the multipulse excitation. The excitation was first segmented into individual pitch periods. For every sequence of n pitch periods, the multipulse pattern from the pitch period in the middle of each sequence was then used to repeat the excitation pattern in the other $n - 1$ surrounding pitch periods. In this process, the overall amplitude envelope of the utterance was left unchanged. We find that as many as 15 periods may be repeated without introducing significant audible distortion in the synthetic speech signal. However, repetition of a single multipulse pattern across all voiced segments of an utterance produced noticeable coloration in synthetic speech. We will discuss further implications of these results in this paper.

2:41

BB9. Optimizing pulse amplitudes in multipulse excitation. Sharad Singhal (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

Multipulse excitation [B. S. Atal and J. R. Remde, Proc. ICASSP 82] is an attractive way of modeling the input to the LPC synthesis filter at low bit rates. In the method described by Atal and Remde, the amplitudes and locations of the pulses are determined in successive stages—one pulse at a time. At each stage, one pulse is determined to achieve maximum reduction in the weighted mean-squared error between the original and synthetic speech waveforms. For closely spaced pulses, successive optimization of individual pulses is inaccurate, and often requires additional pulses to compensate for inaccuracies introduced earlier. This problem can be only partly corrected by reoptimizing all pulse amplitudes at the end. This paper describes a method which locates a pulse at any stage by jointly optimizing its amplitude together with the amplitudes of all pulses located up to that stage. As a result the amplitudes of all pulses are kept optimal at every stage, and very little inaccuracy is introduced even if the pulses are closely spaced. The new method provides significant improvements in the SNR—by as much as 10 dB in some sections of speech.

2:53

BB10. Does a Bernoulli force cause velar double bursts? John Kingston (Phonology Laboratory, Linguistics Department, University of California, Berkeley, CA 94720)

Multiple intensity peaks, "double bursts," are common in the release of velar stops. Keating, Westbury, and Stevens [J. Acoust. Soc. Am. Suppl. 1 67, S93 (1980)] suggest that shortly after the velar release, a rate of air flow arises between the tongue dorsum and the soft palate high enough to pull them back together by a Bernoulli force like that which draws the vocal folds together. The release of this second contact produces a second burst. If a high rate of air flow causes this second, brief closure, then

double bursts should occur more often after /k/ than /g/, since intraoral air pressure during closure is higher for the voiceless stop than the voiced one. Intensity measurements of the initial portions of the bursts of [k^h,sk,g^h,st,d] before [i,ɛ,u] show vowel quality determines burst intensity for velars—least intense before [u], most before [i], with [ɛ] inducing weak to moderate bursts—more than the voiced/voiceless or aspirated/

unaspirated contrasts do. Alveolar releases vary much less across vowels, but peak intensity is reached approximately 10 ms later after [t^h] than [d] or the [t] which follows [s]. Evidently, the point of contact with the soft palate or the movement to the following vowel mechanically determines burst intensity for velars, while for alveolars aerodynamic factors are more potent.

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

DEL MAR/HELIX/SANTA FE ROOMS, 1:30 TO 3:05 P.M.

Session CC. Musical Acoustics IV: The Violin Octet and Other Stringed Instruments

Carleen M. Hutchins, Co-Chairman

Catgut Acoustical Society, Inc., 112 Essex Avenue, Montclair, New Jersey 07042

Bertram Turetsky, Co-Chairman

Music Department, University of California, San Diego, La Jolla, California 92093

Chairman's Introduction—1:30

Invited Paper

1:35

CC1. The violin octet. Carleen M. Hutchins (Catgut Acoustical Society, Inc., 112 Essex Avenue, Montclair, NJ 07042)

Eight new violin family instruments in graduated sizes project the tone quality of the violin into seven other tone ranges, representing the first time a consistent theory of acoustics has been applied to a family of musical instruments. Undertaken in 1956 at the instigation of Henry Brant, they resemble instruments constructed in Europe during the 16th and 17th centuries. Using early violin tests, F. A. Saunders, John C. Schelleng, and C. M. Hutchins ferreted out a controlling characteristic of the violin: namely the two main resonances within a semitone of the two open middle strings. These two resonances are differently placed in viola, cello, and bass. Schelleng and Hutchins developed a mathematical scaling theory and Hutchins spent nearly 10 years adapting and constructing instruments according to these parameters. The first concert was at Harvard in 1963, a memorial to F. A. Saunders; a second in 1967 at the New York YMHA. There are over 50 compositions and arrangements for the OCTET, which has traveled thousands of miles for concerts and lecture demonstrations—one set permanently in England, another in Stockholm, with a composer contest and concert for the "Swedish Musical Acoustics Conference 1983." Violin makers in the USA, Europe, and Australia are constructing the instruments which are described in "THE VIOLIN OCTET" available from the Catgut Acoustical Society. Personal work time privately donated; machinery and acoustical equipment funded by: John Simon Guggenheim Memorial Foundation, Martha Baird Rockefeller Fund for Music, American Philosophical Society, Arnold Hoffman Foundation, George MacDonald Foundation, Estates of Virginia Apgar, Helen Rice, Eunice Wheeler, and the Harriett M. Bartlett Fund of the Catgut Acoustical Society, Inc.

Contributed Papers

2:05

CC2. Violin sound synthesis from first principles. Gabriel Weinreich (IRCAM, 31, rue Saint-Merri, 75004 Paris, France) and Randall Laboratory, University of Michigan, Ann Arbor, MI 48109)

A simple computer model of a violin has been constructed. It consists of two arrays carrying the forward and backward waves on the string, appropriate reflections at the bridge and finger, a bow of finite width, and a combination of parallel two-pole filters to simulate the radiativity of the violin body. The string is assumed to be nondispersive, although an equivalent amount of dispersion can be included in one or both of the boundary conditions. The instantaneous force exerted by the bow on the string is formulated as a single-valued function of their relative velocity. The output signal can be converted into sound. The model has been used to study a number of phenomena, including the dynamic changes in the

Helmholtz motion when the bow velocity changes, and in particular the starting transients.

2:20

CC3. Characterizing guitar sound using time-evolving spectra. E. Paul Palmer and Linda Zuvich (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

The sound of a guitar or similar instrument can be characterized by the time evolution of its partials and its noiselike sound components. This includes the establishment or generation of noise, families of partials, and formants and resonances by the nature of the plucking and the characteristics of the instrument. It particularly includes the decay and interchange of energy among all the sound components. The investigation of these

components is facilitated by using linear and logarithmic-frequency spectral analysis and by using variable-length time windows in acquiring data to be analyzed. Displays of time-evolving spectra demonstrate the different characteristics of classes of guitars such as classical and steel string and the differences between members of each class. The relationships among measurable sound parameters and guitar quality and guitar design and construction will be discussed.

2:35

CC4. An inflatable guitar. Timothy P. White (Journal of Guitar Acoustics, 219 North Main Street, Ann Arbor, MI 48104)

It is readily demonstrated that blocking the soundhole of a steel-string acoustic guitar does not significantly affect the instrument's tone. Wavelength considerations indicate that the high frequencies to which our ears are most sensitive radiate directly from the guitar's soundboard. It is common knowledge among luthiers that, in determining the tone of an instrument, the stiffness and damping properties of the guitar's soundboard are of paramount concern. In this study, the traditional wooden architecture of the guitar's soundboard, typically a spruce veneer reinforced with variable wooden bracing, is replaced on a commercial production instrument with a graphite-epoxy "grill" supporting a thin synthetic polymer membrane under tension. In another instrument, the guitar's traditional wood-

en soundbox is replaced altogether by a graphite-epoxy soundboard grill backed by an inflatable bladder in the shape of a normal guitar body. Both instruments are demonstrated, and techniques for creating aligned-fiber graphite instrument components with room-temperature "time-set" rather than "thermo-set" epoxy resins are discussed.

2:50

CC5. The lids of pianos and harpsichords as acoustic transformers. Edith L. R. Corliss and Charles H. Corliss (Forest Hills Laboratory, 2955 Albemarle Street, NW, Washington, DC 20008)

The lid of a grand piano may be closed altogether on the case, removed, or raised to a fixed position by a stick. Many larger instruments are equipped with a short stick as an alternative to the long stick. Raising the lid to these alternative positions changes the tone color to an extent perceptible to the player and the audience. This effect is measured more readily with a harpsichord because the plucking action is essentially independent of the player's touch. Using a harpsichord built by C. Randall Taylor in 1966, the authors have investigated the effect of adjusting the lid to a number of positions. The change produces measurable alterations in the output. The effect is similar to horn loading on a loudspeaker, and will be discussed from this standpoint.

WEDNESDAY AFTERNOON, 9 NOVEMBER 1983

CALIFORNIA ROOM, 12:30 TO 2:35 P.M.

Session DD. Psychological and Physiological Acoustics III, Noise IV, and CCEA: Effects of Man-Made Noise on Animals

William C. Cummings, Chairman

Oceanographic Consultants, 5948 Eton Court, San Diego, California 92122

Chairman's Introduction—12:30

Invited Papers

12:35

DD1. Audiograms for large cetaceans: A proposed method for field studies. Sam H. Ridgway and Donald A. Carder (Code 5142, Naval Ocean Systems Center, San Diego, CA 92152)

In the great majority of cetaceans, the sense of hearing has not been studied. There are no audiograms on any of the great whales. To us, the ABR (Averaged Brainstem Response) technique seems the only way to get such data on the larger species. We have used the ABR in smaller cetaceans such as *Delphinus*, *Tursiops*, and *Delphinapterus* to record audiometric data. Fine wires about the size of a human hair are inserted into the head through a hypodermic needle. Bare tips of the insulated wire over the vertex and mastoid region pick up electrical potentials (electroencephalogram or EEG). The EEG is amplified, filtered, and tape recorded during periods when sound pulses are presented through a hydrophone. We observe the ABR by synchronous averaging of the EEG for 10 to 25 ms after each pulse. The sound intensity and frequency are varied to determine the threshold at each frequency, thereby producing an audiogram for the animal. Audiometric data would be helpful in assessing the impact of human generated acoustic events on whales. We have developed a portable system that we hope will enable us to make measurements on whales trapped in fishermen's nets, cetaceans stranded in tidal basins, or animals that beach themselves.

12:50

DD2. Compatibility of ships and whales in Glacier Bay, Alaska. James H. Johnson (National Marine Mammal Laboratory, National Marine Fisheries Service, 7600 Sand Point Way N.E., Seattle, WA 98115)

Glacier Bay, southeast Alaska, normally a summer feeding ground for humpback whales, was abruptly abandoned in midsummer 1978, prompting charges by environmentalists that increased vessel traffic had driven them out. Subsequent research focused on whale-prey relationships, vessel effects on whales, and the overall acoustic environment of Glacier Bay compared to nearby areas where whales continue to feed in undiminished numbers. Prey studies showed a high correlation between whale distribution and prey abun-

dance; in 1982 a group of whales chose to spend most of the summer feeding in the busiest, noisiest part of the Bay. Behavior studies showed whales are affected by vessels up to 3-km away, but reactions were of short duration, and meaning in terms of possible physiological stress and cumulative effect is unknown. Acoustic studies revealed minor differences between the Bay and comparison areas (e.g., "white sound" from effervescence of melting glacier ice, and sporadic seismic noise), which presumably are longstanding phenomena that do not explain the recent reduction of whales. Humpbacks are gradually reoccupying the Bay, possibly the result of current traffic regulations and restrictions on numbers of vessels allowed entrance. However, a "crash" of whale prey stocks in 1978 still cannot be discounted. [Work supported by National Park Service, National Marine Fisheries Service, and Alaska cruise ship industry.]

Contributed Papers

1:05

DD3. Behavioral and physiological responses of captive beluga whales to playbacks of drilling noise. Jeanette A. Thomas and Ronald Kastelein (Hubbs Sea World Research Institute, 1700 South Shores Road, San Diego, CA 92109)

There has been recent concern that noise from drilling platforms may have detrimental effects on marine animals. Unfortunately, little data are available on the hearing capabilities of marine mammals and their responses to man-made noise. The objectives of our study were to playback underwater recordings of drilling noise to captive beluga whales and assess: (1) behavioral responses such as respiration rate, dive interval, changes in social combinations, and change in usage of areas in the pool, and (2) physiological responses monitored by changes in blood chemistries, especially catecholamine levels. Baseline behavioral and blood values were collected for two males and two females housed at Sea World, San Diego for a 30-day period. Playbacks were conducted for 9 days preceded by pretest observations. The playbacks were recordings of SEDCO 708 platform and synthesized drilling noise projected at about 150 dB re: 1 μ Pa. The manner in which behavioral parameters and blood chemistry values changed during the playbacks of drilling noise will be discussed.

1:15

DD4. Behavioral responses of wild beluga whales (*Delphinapterus leucas*) to noise from oil drilling. Frank T. Awbrey and Brent S. Stewart (Hubbs Sea World Research Institute, 1700 South Shores Road, San Diego, CA 92109)

Two seasons of field observations and playback experiments with southwest Alaskan beluga whales has established that wild whales in their natural habitat respond more negatively to sudden changes in sound level than to sustained sounds. Recordings of noise from SEDCO 708 drilling platform were projected, underwater at 163 dB re: 1 μ Pa source level. Whales less than 1.5 km from the source usually reacted to onset of the noise by swimming away. In two instances, groups of whales that were at least 3.5 km from the noise source when playback started continued to approach. One group approached within 300 m before all or part turned back. The other group submerged and passed within 15 m of the projector. Results were consistent with observations of captive belugas.

1:25

DD5. Sound and vibration levels in a ringed seal lair from seismic profiling on the ice in the Beaufort Sea. D. V. Holliday (Tracor, Inc., 9170 Chesapeake Drive, San Diego, CA 92123), W. C. Cummings (Oceanographic Consultants, 5948 Eton Ct., San Diego, CA 92122), and D. E. Bonnett (Tracor, Inc., 3505 Anderson Hill Road, Silverdale, WA 98383)

The ringed seal (*Phoca hispida*) habitat on Alaska's North Slope is subjected to intensive local seismic exploration. During the winter and spring months, this seal constructs and maintains lairs for resting and birthing on the ice surface under a snow cover. Measurements of airborne sound and vibration levels in a lair and underwater sound levels beneath the lair are reported as a function of range to a seismic sound source (Vibroseis[®]) operating on the ice near Reindeer Island, north of Prudhoe Bay, AK. Propagation losses for underwater sound in the band between 10 and 70 Hz were from 10 to 20 dB per doubling of range. Vibration

energy losses range between 20 and 40 dB per range doubling for the same frequencies. Losses increased with increasing frequency. [Work sponsored by NOAA's National Ocean Survey, Office of Oceanography and Marine Services.]

1:35

DD6. Acoustic testing procedures for determining the potential impact of underwater industrial noise on migrating gray whales. Charles I. Malme and Paul R. Miles (Bolt Beranek and Newman Inc., 10 Moulton Street, Cambridge, MA 02238)

A procedure was developed for observing the track patterns and behavior of migrating gray whales off Soberanes Point south of Monterey, California. A series of acoustic playbacks were made at realistic levels from an anchored vessel to simulate the presence of oil and gas development and production activities. In another series of tests, a seismic survey vessel with a 4000-in.³ air-gun array was navigated at progressively decreased ranges from the migrating whales. This was followed by a test sequence using a vessel with a single 100-in.³ air gun. Source characteristic and acoustic transmission loss measurements were made to permit prediction of the noise exposure for the nearby subject animals. Exposure levels for which observed behavioral changes occurred were determined. [Work supported by Dept. of Interior, Minerals Management Service.]

1:45

DD7. Effects of underwater noise on migrating gray whales off the coast of California. Christopher Clark (Brockefeller University, Tyrrel Road, Millbrook, NY 12545), Peter Tyack (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), James Bird (16 Mead St., Apt. 52, Allston, MA 02134), and Victoria Rowntree (Weston Road, Weston, MA 01773)

The potential effects of man-made underwater noises on the behavior of gray whales (*Eschrichtius robustus*) off the Coast of central California were studied during their southern and northern migration in 1983. In January, two shore-based theodolite tracking sites recorded whales' swimming patterns as well as behavioral activities while underwater sounds (production platform, drilling platform, semisubmersible, drillship, helicopter, or killer whale, *Orcinus orca*) were transmitted from a vessel anchored in the middle of the nearshore migratory pattern. In April and May, when the majority of whales were mother-calf pairs, three theodolite tracking sites recorded swimming patterns, behavior, and respiration times for individuals. Experimental noise sources consisted of either a seismic vessel operating a 40-gun array, a vessel operating a single air gun or a stationary vessel broadcasting either drillship or killer whale sounds. The behavioral measures used to assay disturbance included deflection of normal migratory pattern, speed of travel, individual respiration, synchronizing of mother and calf respiration, and rates of behaviors such as breaching or underwater expiration. [Work was supported by Minerals Management Service.]

1:55

DD8. Responses of gray whales (*Eschrichtius robustus*) to nonbiological noise. Marilyn E. Dahlheim (Department of Zoology, University of British Columbia, Vancouver, BC, Canada V6T 2A9 and National

Marine Mammal Laboratory, National Marine Fisheries Service, 7600 Sand Point Way N.E., Seattle, WA 98115) and H. Dean Fisher (Department of Zoology, University of British Columbia, Vancouver, BC, Canada V6T 2A9)

In order to focus on the problem of increasing nonbiological noise in the environment of the gray whale (*Eschrichtius robustus*) and to answer the question as to how this species uses its acoustic signaling most efficiently, a study was undertaken at Laguna San Ignacio, Baja California Sur, Mexico, in February 1983 to determine the responses of gray whales to projected sound sources. A series of underwater "playback" experiments were conducted. The playback included the projection of nonbiological as well as biological sound sources. The acoustical and behavioral responses of the whales were recorded during pretrial, trial, and post-trial playback periods. Sounds were projected back only during the "trial" periods. With the onset of increased levels of man-made noise, the gray whale's acoustical activities increased, as did the source levels of their calls. Observations on the behavioral activities of these whales in response to increased noise resulted in (1) prolonged dive times and (2) direct movements on the part of the whales away from the transducers. The acoustical and visual responses of the whales appear to differ depending upon the sound stimuli being presented.

2:05

DD9. Preliminary measurements of sound attenuation by snow over a model seal lair. 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)

Because of their winter habit of denning in snow above Arctic ice, ringed seals, *Phoca hispida*, may be exposed to airborne man-made noise. Incidental to acoustical studies of winter seismic exploration [D. V. Holliday *et al.*, J. Acoust. Soc. Am. Suppl. 1, this meeting] a series of measurements was undertaken of airborne tones received in a model seal lair. Five successive layers of snow blocks were added around the sensor. Seven tones from 105 to 10 510 Hz were played from a loudspeaker at two distances, 0.8 and 11.5 m, through the accumulative five layers (maximum thickness, 0.9 m). This study was undertaken off Prudhoe Bay, Alaska, April 1983. The results indicated an increasing transmission loss through the "lair" wall of about 6 dB/doubled thickness, an increasing dependence with frequency (Hz), and evidence of significant propagation through the ground path. It is tentatively concluded that, in this situation, snow may be an effective barrier for higher frequency airborne noise, but attenuation is limited by flanking propagation through the underlying medium. [Work sponsored by NOAA's National Ocean Survey, Office of Oceanography and Marine Services.]

2:15

DD10. Source levels of bowhead whale sounds determined by acoustical array localization. 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)

One effect of man-made underwater noise is possible masking or other interference with sounds of whales requiring knowledge of the source level. In conjunction with a technical feasibility study of acoustical location in northern Alaska [W. C. Cummings, Tracor Doc. No. T-83-06-002-U (1983)] numerous whale sound source levels were determined based on accurate distances using a three-hydrophone array baseline of 2.4 km. In terms of dB *re*: 1 μ Pa at 1 m (peak spectrum level over the duration of each signal), 24 localized song elements ranged from 158 to 189, median 176.5 dB. The 33 localized moans ranged from 129 to 170, median of 158 dB. At a distance of 10 km, this implies a theoretical received signal-to-noise ratio for songs of 36 dB, based upon the observed median ambient level at 500 Hz, or 16 dB at maximum noise level. However, based upon our experience with the prevailing shallow water propagation losses and noise characteristics, the S/N ratio of songs can be expected to approach 0 at about 15 km, with moans, 5–10 km. [Work sponsored by the North Slope Borough.]

2:25

DD11. Interaction between offshore geophysical exploration activities and bowhead whales in the Alaskan Beaufort Sea, Fall 1982. Donald K. Ljungblad (Naval Ocean Systems Center, Code 5131, San Diego, CA 92152)

Out of concern about the potential effects of marine acoustic geophysical survey work on westward-migrating bowhead whales (*Balaena mysticetus*), the U.S. Minerals Management Service (MMS), in consultation with the U.S. National Marine Fisheries Service (NMFS), implemented a program for monitoring and regulating such work in the Alaskan Beaufort Sea during 1981 and 1982. In 1982 a twin-turbine, high-wing aircraft was used to survey systematically blocks covering approximately 1400 km² near actively "shooting" seismic survey vessels. Direct visual observation was supplemented by the use of sonobuoys to listen to and record underwater sounds made by vessels, air guns, and whales. In addition to the systematic surveys, sustained behavioral observations of bowheads were made on an opportunistic basis, with the objective of identifying possible differences in behavior between whales exposed to seismic sounds and whales not exposed to seismic sounds. No major changes in whale behavior (e.g., flight reactions) were observed. Tests of statistical significance were applied to data on number of blows per surfacing, mean blow interval per surfacing, surface times, and dive times. Only mean surface time of "adults" (i.e., all whales other than cows and calves) in the presence and absence of seismic sounds (1.67 \pm s.d. 0.85 min and 1.36 \pm s.d. 0.59 min, respectively) was statistically significant ($t = 1.988$, $df = 89$, $p < 0.05$). No statistically significant differences were detected for other behavioral parameters in the presence and absence of seismic sounds.

Session EE. Physical Acoustics IV and Underwater Acoustics IV: Acoustical Properties of Porous Media

Thomas J. Plona, Chairman
Schlumberger-Doll Research, P. O. Box 307, Ridgefield, Connecticut 06877

Chairman's Introduction—8:00

Invited Papers

8:05

EE1. Acoustic properties of porous media. David Linton Johnson and Thomas J. Plona (Schlumberger-Doll Research, P. O. Box 307, Ridgefield, CT 06877)

The long-wavelength acoustic properties of two-component composites is reviewed and it is shown that in order for such a system to have two distinct longitudinal modes, one component must be a fluid, the other a solid, and each must form its own percolating cluster. The Biot theory, which treats the two displacements on a separate and equal footing, is shown to give an excellent description of a wide variety of such systems which include: the diffusive mode in polymer gels, 4th sound in superfluid/superleak systems, the diffusion of a fluid pressure pulse through a porous solid, and the additional "Slow Wave" recently observed in a variety of water-saturated porous materials. It is shown how one can independently measure all the necessary input parameters and successfully calculate the effect of fluid saturation on the measured speeds. The essential connection between the acoustic properties and the electrical properties of porous solids is made. Finally, it is shown that, in general, standard multiple scattering theories within the single site approximation are intrinsically incapable of giving even a qualitative description of the acoustic properties of such bipercolating systems; the exception is when one component does not percolate as occurs, for example, in a suspension.

8:30

EE2. Marine sediment acoustics. Robert D. Stoll (109 Oceanography, Lamont-Doherty Geological Observatory, Palisades, NY 10964)

The prediction of velocity and attenuation in acoustic waves which propagate through marine sediments is of key importance in many surveillance and exploration techniques. A mathematical model of the sediment must reflect the influence of such parameters as porosity, overburden stress, particle size, and lithification. The general theory of Biot has been used to construct a model which has been successful in estimating propagation characteristics in a wide variety of sediments. Two kinds of viscous damping are included in the model. One deals with the overall motion of the interconnected fluid field with respect to the skeletal frame. The other accounts for local fluid motion near the grain contacts which produces a viscoelastic response similar to the "squeeze film" phenomenon in lubrication theory. Recent experimental data for sandstone, sand, silt, and other sediments show that the various forms of viscous damping produce a frequency dependent Q in virtually all water-saturated sediments. New data are presented for sand and silt in the frequency range of 2–1500 Hz which demonstrates the effects of both kinds of viscous damping. [Work supported by ONR.]

8:55

EE3. Internal friction and modulus dispersion in fluid saturated porous rock. B. R. Tittman, J. R. Bulau, and M. Abdel-Gawad (Rockwell International Science Center, Thousand Oaks, CA 91360)

In the exploration for gas and oil, the interpretation of seismic/acoustic well logging data requires an understanding of the interaction between acoustic waves and rock under *in-situ* conditions, i.e., combined effects of pressure, temperature, and pore fluid. Our recent lab studies have revealed a significant internal friction peak and an associated modulus dispersion in coconino sandstone when the frequency (88–550 Hz) and/or fluid viscosity (0.0025–13.5 poise) is varied at elevated effective pressures to 3.4×10^7 Pa. The results are consistent with a linear relaxation mechanism involving the flow of a viscous intergranular fluid with a characteristic time proportioned to fluid viscosity. The liquid "squirt"-type model of O'Connell and Budiansky (1977) predicts a relaxation peak at the value of frequency-viscosity observed in this study. The presentation will discuss the experimental approach, the data, their interpretation, and their application to exploration seismology. [Work supported by Consortium of Petroleum Companies.]

9:20

EE4. On the drag and virtual mass coefficients in Biot's equations. A. Bedford, M. Stern, and R. D. Costley (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

Consider a hypothetical experiment in which the solid constituent of a fluid-saturated porous medium is subjected to a uniform oscillatory motion. Biot's equations can be solved for the drag and virtual mass coeffi-

cients in terms of the resulting oscillatory motion of the fluid. Assuming the pores to be cylindrical, the motion of the fluid has been determined by subjecting the wall of a cylinder of viscous compressible fluid to a uniform oscillatory motion and averaging the resulting fluid displacement over the volume of the cylinder. Motions parallel to and normal to the axis of the cylinder have been considered. In the case of motion parallel to the cylinder axis, the obtained coefficients are equivalent to those derived by Biot [J. Acoust. Soc. Am. **28**, 179-191 (1956)] and Hovem and Ingram [J. Acoust. Soc. Am. **66**, 1807-1812 (1979)]. By superimposing the parallel and normal cases, the coefficients for cylindrical pores at an arbitrary angle to the propagation direction have been obtained. Then, by averaging with respect to the angle, the coefficients have been determined for a material containing pores of random orientation. [Work supported by the ONR.]

Contributed Papers

9:45

EE5. Theoretical and experimental study of porous absorbing material. J. -L. Berry, J. Nicholas, and B. Malric^{a)} (Département de génie mécanique, Université de Sherbrooke, Sherbrooke, Canada, J1K 2R1)

Porous ceramic has raised considerable interest given the great number of applications of its absorbing and insulating properties, both indoors and outdoors. Numerous experiments with a Kundt tube enable us to evaluate the sound absorbing characteristics of ceramic samples with variable flow resistivities (5-100 cgs), thickness (20-90 mm) and porosities (60%-80%). The most important factors were shown to be flow resistivity and thickness. Subsequently, a theoretical analysis was carried out in terms of normal incidence absorption coefficient using the following impedance models: Delany and Bazley, Beranek, Cremer, and more recently Attenborough. Material thickness and the presence of rear air space (or other material) are also taken into account in order to simulate more accurately practical situations. It was found that Cremer's approach adapted empirically to ceramics gives the best prediction. Thanks to this study, the optimal flow resistivity and thickness may be determined to obtain the best cost/performance ratio. ^{a)} Also at C.R.I.Q., Ste.-Foy, Quebec, Canada G1V 4C7.

9:55

EE6. Grain contacts and viscous relaxation. William F. Murphy, III, Kenneth W. Winkler, and Robert L. Kleinberg (Schlumberger-Doll Research, P. O. Box 307, Ridgefield, CT 06877)

A micromechanical model is developed for acoustic attenuation and dispersion in sedimentary materials such as sandstones. The theory describes the response of two grains in contact to small sinusoidal loadings. Surface energy and fluid saturation are included explicitly. Grain surfaces are microscopically rough and irregular. We postulate that contact between grains forms numerous small solid-solid contacts and that narrow interconnected gaps remain between the surfaces. As the grains oscillate, liquid must be squeezed out of and sucked back into the gaps. The contact relaxation under normal force is hydrodynamic. The theory offers a unified explanation of several heretofore apparently unrelated experimental observations. The resulting equations predict the stiffness and loss as a function of frequency, effective pressure, fluid adsorption, saturation, viscosity, and temperature. Insofar as the micromechanical predictions relate to continuum acoustic properties, the agreement with observation is excellent.

10:05

EE7. Effects of pressure and fluid saturant on ultrasonic scattering in glass beads. Kenneth W. Winkler and William F. Murphy, III (Schlumberger-Doll Research, P. O. Box 307, Ridgefield, CT 06877)

Compressional wave phase velocity and attenuation have been measured in packings of unconsolidated glass beads as functions of external confining pressure. Samples were studied when vacuum dry and when saturated with different pore fluids. Pressure/pore fluid combinations were chosen to minimize nonscattering loss mechanisms. Phase velocity exhibits negative velocity dispersion and attenuation is proportional to the fourth power of frequency. Increasing confining pressure (to 15 MPa)

significantly reduces the scattering losses (by an order of magnitude at a given frequency in dry glass beads). Decreasing pore fluid compressibility produces a corresponding decrease in the scattering losses. This work shows that theoretical models based on suspensions are not adequate when the grains are interacting to form a solid frame. The present data, along with previous data for fused glass beads, suggest that the scattering strength is related to the contrast between the acoustic properties of the composite and of the grains.

10:15

EE8. Estimation of the components of anisotropic permeability from observations of acoustic anisotropy. Wm. Mansfield Adams (Hawaii Institute of Geophysics, University of Hawaii, Honolulu HI 96822), Stephen W. Wheatcraft, and John W. Hess (Desert Research Institute, University of Nevada, Reno, NV 89506)

Decisions on positioning of monitoring wells in the vicinity of hazardous waste sites can be significantly affected by knowledge of the natural anisotropic hydraulic conductivity. That the vertical permeability is usually less than the horizontal permeability is well known; however, the anisotropy in the horizontal plane may influence the choice of positions for those monitoring wells that are downgradient from the waste-management area. Very little information on the horizontal anisotropy is available. Furthermore, field measurements to estimate the horizontal anisotropy are expensive because at least three interacting wells are necessary, with more than three being preferred. The Kozeny equation suggests that directional anisotropy of the electrical conductivity of a porous media should correlate closely with the anisotropy of the permeability [S. J. Pirson, *Geological Well Log Analysis* (Gulch Publ. Co., 1981), p. 135]. A close relationship should also exist between the acoustic anisotropy and the anisotropy of the permeability. Recent advances in techniques for measuring acoustic velocities in a borehole make it feasible to obtain estimates of the anisotropy of the permeability from observations of the anisotropy of the seismic velocities. [Work supported by EPA.]

10:25

EE9. The absorption of sound by anisotropic porous layers. Shawn Burke (Dept. of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

An analytical model which describes the effect of a directional anisotropy of acoustical properties in porous sound absorbers is presented. The oblique incidence impedance of an anisotropic porous layer is derived in terms of the material's flow resistance, effective mass, and phase velocity measured in two directions. The impedance boundary condition is shown to be strongly dependent upon the absorber's transverse complex velocity, as well as on the angle of incidence. Values of the impedance and oblique incidence absorption coefficient for a layer of Fiberglas are calculated using experimental anisotropic parameter data. The random incidence sound absorption coefficient is then calculated for three boundary condition models of the Fiberglas, over frequencies ranging from 300-1200 Hz. The anisotropic model predicts absorption coefficients that agree better with reverberation chamber data than either an isotropic bulk reacting model or a locally reacting model. The absorption coefficient is shown to be insensitive to weak anisotropies.

10:35

EE10. Sound propagation in HeII with partially locked normal component. Steve Baker, Daniel Rudnick,¹⁾ and Isadore Rudnick (Department of Physics, UCLA, Los Angeles, CA 90024)

We consider the consequences of introducing a flow resistance, R , into the superfluid equations of HeII. We find: (1) The dimensionless quantity that characterizes flow resistance in HeII is $R/\rho_s \omega \equiv r$. (2) In a superleak there is a continuous transition from first to fourth sound and from second sound to a diffusive thermomechanical (TM) wave as r increases from 0 to ∞ . (3) In a pressure released system (such as a shallow puddle of HeII on a flat substrate) there is a continuous transition from second sound to a TM surface wave and from gravity wave to a diffusive TM surface wave as r increases from 0 to ∞ . (4) For each system the attenuation per wavelength for the propagating mode is a maximum near $r = 1$, decreasing monotonically as $r \rightarrow 0, \infty$. [Work supported by ONR.]²⁾ Present address: Scripps Institute of Oceanography, USCD, LaJolla, CA 92093.

10:45

EE11. Acoustic scattering correction in a superleak partially filled with superfluid helium. S. Baker, J. Marcus, G. Williams, and I Rudnick (Department of Physics, University of California, Los Angeles, CA 90024)

The index of refraction n has been measured for the sound mode which propagates in a superleak partially filled with HeII so that the liquid forms a film on the powder grains. The mode is a thickness wave in this film. Motivated by a model which treats the scattering from the powder grains independently from the scattering from the vapor spaces, we determine the values of beta in the equation, $n^2 = f^{-\beta_1} P^{-\beta_2}$, where f is the fraction of the pore volume filled with liquid HeII and P is the fraction of the volume occupied by both liquid and vapor ⁴He. The values of beta are interpreted in terms of the tortuosity of the microscopic geometry. The scattering correction has been measured for $0.3 < f < 1.0$. In this range the agreement between the data and the model is excellent with values of $\beta_1 \approx 0.71$ and $\beta_2 \approx 1.3$. When $f = 1$, the pores are filled with liquid and the system is a fourth sound resonator. The value of n^2 is in excellent agreement with directly measured fourth sound values. [Work supported by ONR.]

10:55

EE12. Coupling of airborne sound into the porous elastic surface of the Earth. Henry E. Bass, Lee N. Bolen, and James M Sabatier (Physical Acoustics Laboratory, The University of Mississippi, University, MS 38677)

The surface of the Earth can be modeled as an elastic medium permeated with pores. This porous nature has enabled prior investigators to explain measurements of sound propagation in the vicinity of the Earth by describing the surface as a locally reacting boundary. Physically, one can envision the pressure variations associated with sound wave passage forcing air flow in the pores. Some of this energy is lost to the pore walls due to viscous drag and thermal conduction. Viscous drag causes the solid skeleton to move giving rise to a seismic wave in the frame. This wave has been measured with geophones. This physical description conforms well to Biot's description of sound waves in a porous medium and that theory has been used to model the coupling process. Specifically, the complex propagation constants have been calculated using a code developed by Attenborough which includes pore tortuosity and these propagation constants have been used to compute geophone and buried microphone response in a porous medium below an air half space and above an elastic lower half space. Computer response will be compared with measurements reported earlier. [Work supported by ARO.]

11:05

EE13. Propagation of intense sound in fibrous bulk porous materials. D. A. Nelson, D. T. Blackstock (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78712-8029), and N. D. Pereira (Department of Mechanical Engineering, The University of Texas at Austin, TX 78712)

The work reported here is an extension of that by Kuntz [NASA CR 167979, September 1982; see also J. Acoust. Soc. Am. Suppl. 1 69, S80 (1981)], who investigated the nonlinear behavior of intense sound propagating through air-filled bulk porous materials. The primary purpose of the current work is to explain some of Kuntz's experimental observations. Our model is based on the assumption that the nonlinear flow resistivity of the material is far more important than the ordinary hydrodynamic nonlinearities that affect an open fluid. It is further assumed that the nonlinear resistance term is proportional to either $u^2 \text{sgn}(u)$ or u^3 , where u is particle velocity. Both forms lead to harmonic distortion of the "cubic" type. In this case the distortion components of an initially sinusoidal wave are exclusively odd harmonics. This and other predictions are supported by Kuntz's data. The model is thus verified. [Work supported by NASA and ONR.]

11:15

EE14. A comparison between Biot model predictions and shear wave attenuation measurements in unconsolidated laboratory sediments. Burlie A. Brunson (Planning Systems Incorporated, 7900 Westpark Drive, Suite 600, McLean, VA 22102)

Shear wave attenuation measurements were made using three different water-saturated sediments. The sediment assemblages used in the attenuation measurements included well-sorted spherical glass beads, a well-sorted angular quartz sand, and a moderately sorted angular quartz sand. The median grain size of the sediments was held constant while the grain shape and size distributions were varied. Sediment physical properties were measured under the same conditions prevailing during the acoustic measurements. The properties were used as inputs to a theoretical attenuation model based on the Biot theory of propagation of waves in porous media. The model allowed attenuation versus frequency predictions to be made for each of the three sediment assemblages over the measurement frequency range of 1-20 kHz. The resultant comparisons between the measured and predicted attenuations demonstrated the importance of using measured model inputs obtained under controlled laboratory conditions when theoretical model capabilities are being evaluated. The model comparisons shed significant light on the ability of this particular model to predict shear wave attenuation in nonideal sediments. [Work supported by NORDA.]

11:25

EE15. Grain shape and sorting effects on shear wave attenuation in unconsolidated laboratory sediments. Burlie A. Brunson (Planning Systems Incorporated, 7900 Westpark Drive, Suite 600, McLean, VA 22102)

Shear wave attenuation measurements were made using ceramic bimorph transducers to excite transverse vibrations in a cylindrical column of unconsolidated sediment. Three different water-saturated sediments were used in an attempt to determine the effects of grain shape and sorting on the frequency dependence of attenuation. The mean grain size of the sediments was held constant while the grain shape and size distributions were varied. The sediment assemblages used in the attenuation measurements included a moderately sorted angular quartz sand, a well-sorted angular quartz sand, and well-sorted spherical glass beads. The moderately sorted sand showed the greatest attenuation over the measurement frequency range of 1-20 kHz. The well-sorted sand and the glass beads showed generally lower attenuation with the beads being the least lossy propagation medium. All three sediments showed evidence of viscous attenuation due to fluid-to-grain relative motion. This mechanism leads to a nonlinear relationship between attenuation and frequency. [Work supported by NORDA.]

11:35

EE16. Effects of the hydraulic anisotropy and the elastic anisotropy of a porous bottom on the acoustic propagation in shallow water. Tokuo Yamamoto and Mohsen Badiy (Division of Ocean Engineering, Rosenstiel School of Marine and Atmospheric Science, University of Miami, Miami, FL 33149-1098)

The layered anisotropic porous bottom was modeled by the stress-displacement matrix derived from the Biot equations for the anisotropic

porous media. The effects on the acoustic mode attenuation of the anisotropy in the permeability and the elastic moduli of sediments were calculated for the realistic range of the physical properties of naturally consolidated sand deposits and the frequency range of 10–3000 Hz. It was found that the anisotropy in permeability has a significant influence on the acoustic mode attenuation while the anisotropy in the elastic moduli has a small influence. [Work supported by ONR.]

11:45

EE17. Angle-resolved spectroscopy of fluid-filled porous graphite. J. F. Muratore and Herbert R. Carleton (College of Engineering and Applied Science, SUNY at Stony Brook, Stony Brook, NY 11794)

We have investigated the ultrasonic properties of water-saturated porous graphite as a planar sample is rotated in the field of an ultrasonic beam over the frequency range of 0.1–10 MHz. Amplitude spectra are calculated for a range of rotational angles and show effects due to reflection and refraction of the beam, mode conversions, and extensive attenuation and dispersion in the bulk materials. Corrections due to the finite width and diffraction of the beam are to be taken into account to show any shifts in time delay that might affect the accuracy of the data. Energy conversion to a shear mode at "normal" incidence is clearly demonstrated in the time domain signals and shown to account for shifts in apparent amplitude of the longitudinal mode. Our results demonstrate the effectiveness of digital signal processing and selective windowing in FFT analysis in studying the different mechanisms involved in the behavior of such highly dispersive materials subject to elastic wave propagation.

THURSDAY MORNING, 10 NOVEMBER 1983

COUNCIL/CHAMBER ROOMS, 9:00 A.M. TO 12:20 P.M.

Session FF. Engineering Acoustics IV and Musical Acoustics V: Harry F. Olson Memorial Session

Harry B. Miller, Chairman

Code 3234, Naval Underwater Systems Center, New London, Connecticut 06320

Chairman's Introduction—9:00

Invited Papers

9:05

FF1. Some personal recollections of early experiences on the new frontier of electroacoustics during the late 1920s and early 1930s. Frank Massa (Massa Products Corporation, Hingham, MA 02043)

During this brief presentation, time will be turned back a half-century to permit some personal recollections of early experiences as we explored the new frontiers in electroacoustics. The availability of the vacuum tube for the amplification of weak signals made possible the practical application of a variety of wide-range low-sensitivity transduction techniques for the generation and reproduction of sound. Our training in electrical engineering provided an understanding of electrical measurements and design techniques that we could adapt for new specialized uses during our early stages of acoustic measurements and electroacoustic research. The demands of the new talking picture industry for improved electrical recording and reproduction of sound helped finance the costs of electroacoustic research which resulted in rapid progress in the sound reproduction and recording industry during the early 1930s. In less than a decade the reproduction of sound was transformed from the primitive limited frequency range of the mechanical phonograph to the high quality electrical recording and reproduction made possible by the rapid advances in electroacoustic engineering.

9:35

FF2. RTSKED, a real-time scheduled language for controlling a music synthesizer. M. V. Mathews (Bell Laboratories, Murray Hill, NJ 07974)

H. F. Olson helped create the RCA synthesizer. This machine was the first to use sophisticated digital control. Control data were punched into a wide roll of paper tape using a binary code. New technology has superseded paper tape, but control remains the principal problem for modern synthesizers. Precise control of the timing of events in music is difficult for stored program computers. A control language is described in which the computer executes a schedule. A schedule differs from a normal program in that commands and the times at which these commands are executed are separately specified, thus making a clean separation between what the computer does and when it is done. Times can be specified in absolute terms (wait so many milliseconds) or be specified in relative terms (wait until a performer presses a particular key). The language is intended to control real-time performance. It allows the flexible combination of information from a score in the computer memory with information generated by the performer playing on a keyboard or on other sensors. Early experiments indicate that RTSKED is easy to learn, pleasant to use, and powerful.

10:05

FF3. Theory, ingenuity, and wishful wizardry in loudspeaker design—A half-century of progress? George L. Augspurger (Perception Inc., Box 39536, Los Angeles, CA 90039)

During the past 50 years the course of professional loudspeaker design has been well-documented. The history of high-fidelity consumer products is less familiar and more erratic. Loudspeaker designers seem to

have embodied three archetypes in varying proportions: the elegant theoretician, the inspired tinkerer, and the wishful wizard. A brief examination of some of their enduring successes, ephemeral favorites and resounding failures is presented.

10:35

FF4. Some remarks on electro-mechano-acoustical circuits. Leo L. Beranek (7 Ledgewood Road, Winchester, MA 01890)

The advanced development of electro-mechano-acoustical circuits dates in the USA from Firestone (1933), Mason (1941), Olson (1943), Le Corbeillier and Yueng (1952), Bauer (1953), and Beranek (1954). Although the connection between the equations of radio and acoustic wave motion was clearly recognized by Helmholtz and pointed up by Rayleigh, the use of lumped-element circuits was brought on by the advent of radio. Significant use of electrical (radio) analogies was made by Darrieus (France, 1929) and Haehnle (Germany, 1932). This paper treats the period in which Olson's work was centered and cites some examples of product development that were influenced by the concepts of electro-mechano-acoustical circuits.

11:05

FF5. Magnetic recording, past and present. Marvin Camras (IIT Research Institute, 10 West 35th Street, Chicago, IL 60616)

Why is the old complaint true: "The ancients have stolen our inventions?" There are good reasons: It is easy to propose something, difficult to make it work. Excellent ideas are often discarded when a trifling mistake gets negative results; later a luckier discoverer gets the credit. Case histories in magnetic recording illustrate these points. Metal particle tape was proposed 95 years ago. High frequency bias had its roots in the Marconi-Muirhead wireless detector of 1902. Digital recording goes back at least to Morse's telegraph of 1840. The rotating-head principle used for video recorders dates back about three decades before it became practical. Sendust used in the latest recording heads was invented in 1936. Magnetic disks, the important memory of personal computers, were demonstrated in about 1905. Soundstripping for motion picture film goes back to the early 1920s. Personal experiences of failures and successes are related.

11:35

FF6. Fifty years of stereo phonographs: A capsule history. C. Roger Anderson (Shure Brothers Inc., Evanston, IL)

From the 45° disk recording system shown in the 1933 benchmark stereo work of Blumlein to the present day generations of pickups with ultrasonic capabilities is a long evolution of analytic and engineering effort. This paper is a recognition of advances along the way, such as the theory of tracing distortion, stylus contours, transducer principles, light and stiff moving systems, tracking requirements, and the design insight that led to contemporary pickups which perform superbly at 1-g stylus force.

11:50

FF7. Engineering highlights of the LP record. Daniel W. Gravereaux (Recording Research, CBS Technology Center, 227 High Ridge Road, Stamford, CT 06905)

For 35 years, the long playing microgroove 33 $\frac{1}{3}$ rpm disk record has brought the highest fidelity programming of any mass media to the consumer. This is due to the valued efforts of many engineers, scientists, and technicians in the development and implementation of microgroove mastering techniques, cutting and playing transducers, electroplating, stamping techniques, lacquer blanks, cutting styli, mastering lathes, presses, and vinyl formulation. This paper presents a few of the highlights in the engineering of the most standardized sound medium in the world—the LP.

12:05

FF8. Reminiscing—The stereophonic record. H. E. Roys (RCA Records, 327 S. Paseo Chico, Green Valley, AZ 85614) [Author will not present paper.]

The author became active in the sound recording field in the early 1930s and remained active in that field until retirement in 1967. When, in 1957, two slightly different systems of recording stereophonic records were introduced for standardization he was a member of the engineering committee of the Record Industry Association of America and Chairman of the Phonograph Committee of the Electronic Industries Association. The situation was unique in that stereophonic recorders and pickups were not readily available and the choice for standardization had to be based largely upon theoretical considerations. The difference between the two was slight. The V-L system recorded one channel vertically and the other laterally. The 45-45 system recorded both channels vertically, but at a 45° angle with respect to the surface of the disk. Record engineers in Europe gathered and decided in favor of the 45-45 system. Engineers in this country did likewise. Thus for the first time in the history of records a common agreement with respect to standardization was reached prior to mass production.

Session GG. Architectural Acoustics V and Noise V: Measurements, Methods, and Materials

Alfred C. C. Warnock, Chairman

National Research Council of Canada, Division of Building Research, Montreal Road, Bldg. M-27, Ottawa, Ontario, Canada K1A 0R6

Chairman's Introduction—9:00

Contributed Papers

9:05

GG1. Spatial transformation of sound fields. Jørgen Hald and Ole Roth (Brüel & Kjær, Nærum Hovedgade 18, DK-2850 Nærum, Denmark)

In this paper it will be demonstrated how cross spectra measured by a modified 1/3 octave realtime intensity analyzer during a planar scan can be used to predict pressure, particle velocity, and intensity in a half-space not containing the source of the (noise) field. We just have to assume a rather limited number of significant uncorrelated components in the total source. In the nearfield region on both sides of the scan plane the field quantities are evaluated using a nearfield holography technique similar to the technique described by Williams, Maynard, and Skudrzyk [J. Acoust. Soc. Am. 68, 340-344 (1980)] for analysis of coherent sound fields. Outside the nearfield region Helmholtz' integral equation is used for the prediction. The combined method allows various simulations to be carried through. For example we can estimate a planar source distribution by means of the holography method, simulate an attenuation of some part of the source, and investigate the effect of this attenuation on the field in the entire source free half-space.

9:20

GG2. A transducer and processing system to measure total acoustic energy density. Marc Schumacher and Elmer L. Hixson (Department of Electrical Engineering, The University of Texas at Austin, TX 78712)

A four-microphone orthogonal array is used to determine the acoustic pressure and the three components of acoustic velocity over the range of 300-3000 Hz. Signals are squared and summed to produce potential, kinetic, and total energy density. The three components of the intensity vector can also be determined. The device has been designed for the study of and improved measurements in reverberant and partially reverberant sound fields. Some preliminary measurements in a reverberation room will be presented. [Work supported by IBM Corp., Communications Products Div., Austin, TX.]

9:35

GG3. Sound intensity errors in a reactive field. Jean Nicolas and Gilles Lemire (Université de Sherbrooke, Génie mécanique, Sherbrooke, Canada J1K 2R1)

During the past years many *in situ* intensity measurements have been conducted by the authors looking ahead for a better understanding of the phenomena involved when quantifying sound power of a particular source in the presence of external perturbing noise. It became evident that the limitations depend on the following question: is the field progressive or reactive? A general formulation is developed for both types of field. In a reactive field, the fact that the instantaneous phase variation of the pressure is different from that obtained for a progressive field is taken into account. This leads to enormous uncertainties that have no common magnitude with errors associated with progressive field. In order to determine these uncertainties when measuring sound power, a model with two or more point sources creating various degrees of reaction on the measurement envelope is used. It shows the action and interaction of the main parameters (power ratios of the sources, distance, number, and spread of measurement points). Interestingly enough it is shown that an increase in the number of points does not reduce the uncertainty in the case of a highly reactive field.

9:50

GG4. Acoustic intensity measurements in the reflective environment. A. Mielnicka-Pate (Department of Engineering Science & Mechanics, Iowa State University, Ames, IA 50011)

The two-microphone intensity method was used to investigate the intensity distribution and radiated power of several sources. The sources used were two open pipes of 30-mm and 95-mm diameter and an aluminum circular plate of 100-mm diameter. The pipes and plate were mounted with baffles. The investigations were confined to the low-frequency range of plane-wave propagation inside a pipe. The experiment was performed in the nearfield and farfield of the sources in the reflective environment. The theoretical and experimental results will be discussed for both intensity and radiated power. The influence of the source impedance will be analyzed for the intensity measurements, particularly at points located close to the source. The intensity error that can occur due to the partial absorption of the reflected waves by the source will be discussed.

10:05

GG5. Effect of the sound field structure on the measurement of acoustic intensity. G. W. Elko and Jiri Tichy (Graduate Program in Acoustics, P. O. Box 30, State College, PA 16801)

The effect of instrumentation phase and magnitude error on the measurement of the active and reactive acoustic intensity for different sound fields will be discussed. We will show experimental results of measurements made for the active and reactive intensity in some semireverberant fields. An easy method to calculate the effects of phase bias will be presented and demonstrated for the experimentally measured data. The measured results will be compared to theoretically predicted values and the measuring system will be described.

10:20

GG6. A portable, low-cost, digital system for room acoustics measurements. Robert D. Essert (Artec Consultants, Inc., 245 Seventh Avenue, New York, NY 10001)

A portable digital acquisition and analysis system has been developed around a low-cost microcomputer. The system consists of a programmable transient signal generator and an acquisition and analysis section. Designed principally for impulse response measurements in both full scale and model rooms, the acquisition system has been configured for portability and ease of use in the field. Data is "captured" directly into the computer main memory and is stored in digital form on floppy disks. A resident graphics module can be used to display the impulse response in various forms and to display the sound energy decay curve computed from the impulse response by backward integration. General purpose signal processing software has been written for computation of Fourier transforms, power spectra, cepstra, and digital filtering. The use of Wiener filtering for the deconvolution of room impulse response and source signal is being implemented, and is the subject of a companion paper. Some results of measurements made in existing performance spaces will be presented.

10:35

GG7. Acquisition and analysis of architectural acoustic data using a microcomputer. E. Paul Palmer and Rodney Price (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

A microcomputer is easily adapted to the task of acquiring up to 32-k samples of 8-bit digitized acoustic data at rates up to 20 k bytes/s. This makes it suitable for acquiring and analyzing architectural data such as impulse response, reverberation time, speech intelligibility, etc. We will compare computer-simulated and experimentally obtained impulse-response data for a simple rectangular room. Comparisons will be made for data acquired using an integrated-signal spark sound source, a pseudorandom-sequence, white-noise source, and a swept-sine source. Difficulties met in using a pseudorandom sequence of length different from a power-of-two will be discussed. Some limitations met in using low-resolution, 8-bit data will be indicated. System variations and improvements will be noted.

10:50

GG8. Influence of reverberation room volume on measured absorption coefficients. A. C. C. Warnock (National Research Council Canada, Division of Building Research, Noise and Vibration Section, Building M-27, Ottawa, Ontario, Canada K1A 0R6)

Recent absorption measurements in four rooms ranging in volume from 16 to 250 m³ suggest that the measured coefficients depend on the logarithm of the room volume; i.e., smaller rooms give smaller coefficients. Similar effects are suggested by round robin measurements in other countries. Absorption coefficients are also found to depend on the decay rate in the reverberation room without the specimen. These results will be presented and their implications with respect to standard tests will be discussed.

11:05

GG9. Best estimates of initial decay rates in reverberation rooms. Richard M. Guernsey (Cedar Knolls Acoustical Laboratories, 9 Saddle Road, Cedar Knolls, NJ 07927)

The *desideratum* in laboratory decay measurements is to get a good estimate of the decay rate of the reverberant field immediately after the source is turned off. When a decay is the superposition of two or more modes with different decay rates the level is not a linear function of time. Thus we want to detect a curved decay and make a best estimate of its initial decay rate. Currently, a real time analyzer controlled by a digital computer provides the best practical measurement of decays. A linear,

first order regression is fitted to the ensemble average of a suitable number of decay measurements. Suggestions by Bodlund; Halliwell and Warnock; and Bartel for detecting curved decays are evaluated. Two additional methods are proposed. It is demonstrated with the help of statistically simulated, curved decays that a second-order, linear regression provides a better estimate of the initial decay rate than a first order regression; the penalty in calculation time is not severe. Guidelines for future investigations are proposed.

11:20

GG10. Comparison of impedance measurement techniques. J. T. Mason, G. B. Mills, S. L. Garrett (Physics Department, Code 61Gx, Naval Postgraduate School, Monterey, CA 93940), and James L. Wayman (Mathematics Department, Naval Postgraduate School, Monterey, CA 93940)

The acoustic impedances of a Helmholtz resonator and porous materials were measured using two different techniques. The first method used the conventional Standing Wave Ratio technique. The second was the two-microphone technique of Chung and Blaser [J. Acoust. Soc. Am. **68**, 907-921 (1980)]. The calculability of the frequency dependent Helmholtz resonator impedance allowed the methods to be compared for both absolute accuracy and realtive precision. [Work supported by the NPS Foundation Research Program.]

11:45

GG11. Effect of sealants on the sound absorption coefficients of acoustical friable insulating materials. Mary K. Lory (Department of Speech and Hearing, University of California at Santa Barbara, Santa Barbara, CA 93106) and James L. Wayman (Department of Mathematics, U. S. Naval Postgraduate School, Monterey, CA 93940)

Acoustical friable insulating materials (AFIM), which often in the past have contained asbestos, have been used for sound control since the mid-1930s. Because of their widespread use and the ease of fiber dissemination, friable asbestos materials are considered to be the major source of asbestos fiber contamination in the indoor environment. Encapsulation of asbestos materials with a commercial sealant product is one of several methods used to control potential asbestos exposure in rooms. A sealant product that preserves most of the acoustical properties of the material is preferred in this usage. In this study, AFIM sample materials were treated with six types of sealants and the effects on normally incident absorption coefficients from 100-2500 Hz were measured using a fixed, dual-microphone technique. "Penetrating" type sealants were found to have a less detrimental effect on sound absorption than those of a "bridging" type.

THURSDAY MORNING, 10 NOVEMBER 1983

SUNRISE ROOM 9:00 TO 11:35 P.M.

Session HH. Physiological Acoustics II: Measurement of the Ear; Evoked Potentials

David L. McPherson, Chairman

Department of Pediatrics, University of California, Irvine, 101 City Drive, Orange, California 92668

Chairman's Introduction—9:00

Contributed Papers

9:05

HH1. Probe measurement of eardrum SPL in occluded real ears. Samuel Gilman, Donald D. Dirks, and David G. Hanson (Department of Head and Neck Surgery, Room 62-129, CHS UCLA, Los Angeles, CA 90024)

Previously, we reported results from simulator studies of techniques for remote measurement of eardrum SPL in hearing aid occluded ears. These data provided means for calibrating remote-measuring probes which included the effects of ear canal length and impedance at the eardrum. The present paper presents data with respect to the accuracy of probe/earmold measurements of eardrum SPL in real ears. Measure-

ments were made on fresh cadaver ears in which the eardrum SPL was independently measured with a micro-miniature microphone placed adjacent to the eardrum. Simulator-derived corrections obtained from the previous study were applied to the probe/earmold measuring system to compensate for ear canal length and assumed eardrum impedance. Results from several ears showed the wide variation associated with individual ear measurements and corrected probe/earmold errors of less than ± 4 dB for frequencies up to 6.0 kHz.

9:20

HH2. Simpler realization of ear simulators using tuning stubs instead of Helmholtz resonators. Juergen Schroeter (Lehrstuhl f. Allg. E-Technik und Akustik, Ruhr-Universität, Postfach 102148, D-4630 Bochum, Federal Republic of Germany)

The Zwislocki- and the IEC711- ear simulators are the widely accepted standard devices whenever a mechanically stable representation of the human ear impedance is needed, e.g., for earphone calibration. Though the design of these simulators is rather complex [see Burkhard, *J. Audio Eng. Soc.* **25**, 1008-1015 (1977) for some related problems], they were not intended to reveal "correct" results above 10 kHz. Recently measurement techniques for obtaining ear impedances data up to 20 kHz have become available [Joswig, *Acoust. Soc. Am. Suppl.* **1** **69**, S14 (1981) and Hudde, *Acoust. Soc. Am.* **73**, 24-31, 242-247 (1983)]. To construct an ear simulator for high frequencies we studied the effect of closed-end stubs of small diameter to replace the lumped-parameter resonators used so far. These stubs do not require any special damping material. The geometrical tolerances are high. An *a priori* error estimation is possible during computer simulation. Predicted and measured results agree well enough to make *a posteriori* fine tuning redundant.

9:35

HH3. Analog filtering of the middle latency auditory evoked response. David L. McPherson and Lance Montgomery (Departments of Pediatrics and Neurology, University of California, Irvine Medical Center, 101 City Drive, Orange, CA 92668)

There has been confusion on the origin of wave P0 of the middle latency auditory evoked response (MLR). It has been suggested wave P0 is a residual of the effects of restrictive filtering of wave V of the auditory brainstem response (ABR). Twelve subjects were used to study the effects of analog filtering on wave V of the ABR, and waves N0 through Nc. Monaural clicks were presented at 70 dBnHL. Two series of 1024 averages were collected for a 100-ms sample for nine filter conditions. High-pass filtering caused greater distortion of the waveforms than did changes in low-pass filtering. When high-pass filtering was at or above 150 Hz, P0 could not be consistently identified. It was observed that filtering of both the MLR and ABR had differential effects on ipsilateral versus contralateral recordings. The later MLR waveforms showed greater variability than either the earlier MLR waves, or the ABR wave V for changes in high-pass filtering, especially above 150 Hz. This study is in agreement with the concept of simultaneous recording of the ABR and MLR reported by Scherg [M. Scherg, *Electroenceph. Clin. Neuro.* **54**, 339-341 (1982)] although that study did not directly address the question of filtering effects.

9:50

HH4. High-pass masker slopes and the BAER. Kurt E. Hecox, Tom Kilsdonk, and Mary Malischke (Waisman Center and the Department of Neurology, University of Wisconsin-Madison, Madison, WI 53706)

The use of analog filters is a nearly universal practice in masking studies of auditory evoked potentials. Yet there is little systematic information on the impact of certain aspects of the filter's properties. This study reports the results of varying masking noise cutoff frequency (8, 4, 2, 1, and 0.5 kHz) and slope (96, 72, 48, 24, 12, and 6 dB/oct), on the response

elicited by a 60-dB HL click. The shallower the filter slope, the longer the latency and smaller the amplitude of the response; the magnitude of this effect interacts with cutoff frequency. There is little evidence of differences between 48 dB/oct and 96 dB/oct slopes, and the maximum change was between 24 and 12 dB/oct for most cutoff frequencies. Since acoustic distortion problems increase with increasing roll-off steepness, it is suggested that 48 dB/oct is an optimum value for brainstem auditory evoked response masking studies. The age dependence of these principles will be shown for newborns. In general, all of the observed latency shifts are consistent with known traveling wave mechanics.

10:05

HH5. The BER, masking, and cochlear place. Robert Burkard and Kurt Hecox (Waisman Center, University of Wisconsin-Madison, Madison, WI 53706)

Two experiments concerning the effects of masking on the brainstem evoked response (BER) are reported. The first experiment evaluated the effects of noise level and rate on waves I, III, and V. There was an increase in peak latencies and a decrease in peak amplitudes with increasing noise level and rate. Higher noise levels and rates increased the I-V interval, and produced a greater increase in the III-V than the I-III interval. The second experiment used the high-pass subtractive-masking technique and covaried derived-response bandwidth and within-band noise level. For half-octave derived bands, the within-band wave V latency at higher noise levels was greater than the unmasked latency of the immediately apical band. The magnitude of within-band noise-induced wave V latency shift was independent of response bandwidth. The relatively large within-band noise-induced wave V latency shift combined with the increasing I-V interval with increasing noise level suggest that a shift in cochlear region of response emanation is not responsible for most of the wave V latency shift with increasing noise level. [Work supported by NIH.]

10:20

HH6. The effects of signal/masker phase conditions on the late cortical response. Amy B. Schaefer, Charles D. Martinez, and Douglas Noffsinger (Audiology and Speech Pathology W126, University of California, Los Angeles and VAMC West Los Angeles, Wilshire and Sawtelle Boulevards, Los Angeles, CA 90073)

Previous investigations have produced indirect evidence of a coherent relationship between the presence and magnitude of the binaural masking level difference (MLD) and the pattern of monaural early auditory brainstem potentials exhibited by the same subjects. This investigation sought electrophysiologic changes corresponding to the presence or absence of behavioral MLD by monitoring via surface electrodes the late cortical potentials (P1, N1, P2, N2) aroused by signal/masker configurations known to produce an MLD. Tonal stimuli of 500 and 2000 Hz were presented binaurally along with binaural narrow-band noise centered at the test frequency under SoNo and S π No signal/masker phase conditions. Twenty-five young (\bar{X} age: 24.6 yrs) normal-hearing subjects with normal ABR for 2000- and 4000-Hz tone pips and normal 500-Hz behavioral MLD were tested. A consistent and statistically significant increase in the amplitude and/or reduction in the latency of onset of the cortical potentials was evidenced for 500-Hz stimuli for the S π No condition when compared to the SoNo condition. Such changes did not occur in response to 2000-Hz stimuli. [Work supported by NIH.]

10:35

HH7. Hemispheric difference in evoked potentials to spatial sound field stimuli. Y. Ando and I. Hosaka (Faculty of Engineering, Kobe University, Kobe, Japan 657)

In order to produce different subjective spatial impressions, two sound field stimuli were alternately presented with a low and a high magnitude of interaural crosscorrelation (IACC). (Subjective diffuseness is the percept for sound fields with the low IACC.) Auditory evoked potentials (AEP) from both temporal areas (T_3 and T_4) of five normal subjects for

each stimulus were recorded. When a bandlimited white noise was presented as a source signal, amplitudes of AEPs over the right cerebral hemisphere were much greater than those over the left, as similar to previous results by several investigators. However, this tendency was only weak, even if a vowel (a) was presented. Furthermore, it is generally found with the two sound signals that (i) for the sound fields with high IACCS, amplitudes of AEPs over the right hemisphere were always greater than those over the left, and that (ii) time differences between the first and the second maxima of AEPs over the right hemisphere were clearly longer than those over the left to the high IACC fields. From these findings, the right cerebral hemisphere is considered to be dominantly operative for the auditory spatial sensation.

10:50

HH8. Human auditory steady state potentials analyzed with Fourier analysis: The zoom technique. R. Dean Linden, Gilles Hamel, David R. Stapells, and Terence W. Picton (Human Neurosciences Research Unit, University of Ottawa Health Sciences, 451 Smyth Road, Ottawa, Canada K1H 8H5)

The Fourier analysis of evoked potentials has several advantages over averaging. Regan introduced a "zoom" technique that manipulates stimulus parameters during the analysis [Invest. Ophthalmol. 12 669-679 (1973)]. Six subjects were tested using this technique to assess the effect of stimulus rate on the auditory evoked potentials. A 95-dB peSPL tone with 4-ms rise and fall times and a 2-ms plateau was presented to the right ear at a rate that increased from 10 to 60 tones/s over 50 s. This analysis was carried out for four different frequencies of the tone: 500, 1000, 2000, and 4000 Hz. The amplitude and phase outputs from the Fourier analyzer were repetitively recorded during the sweep in stimulus rate and averaged over 32 sweeps. The maximum amplitudes of the steady-state responses for the 500, 1000, 2000, and 4000 Hz tones occurred at rates of 40/s, 43/s, 44/s, and 46/s with amplitudes, of 1.42, 1.34, 1.30, and 1.11 μ V, respectively. This technique may be used in "zooming" other parameters of interest such as intensity.

11:05

HH9. Phase consistency of evoked responses to auditory stimuli delivered at high rates. Scott Makeig and Robert Galambos (Children's Hospital Research Center, 8001 Frost Street, San Diego, CA 92123)

We will review our recent research in evoked response detection using auditory stimuli at rates around 40 Hz with emphasis on response phase consistency. Response phase often aggregates at stimulus intensities producing little or no rise in response amplitude beyond the level of the background EEG, and thus promises to be useful in electric response audiometry. We note that others have previously proposed heavily weighting phase statistics in detecting responses to weak stimuli delivered at low rates (around 1 Hz). This sensitivity of evoked response phase to external stimulation suggests that phase of ongoing brain activity may be reorganized by weak stimuli before there is a detectable increase in net response amplitude.

11:20

HH10. Characteristics of the FFP to tonal stimuli. S. B. Resnick, J. M. Heinz, E. W. Bookstein, and R. V. Pagano (John F. Kennedy Institute, 707 N. Broadway, Baltimore, MD 21205)

The FFP to tone bursts of 12.8-ms duration was measured over a 51-ms period using multichannel recording for seven normal-hearing adults. Averaged response waveforms based on 8192 trials were obtained at each of seven frequencies (between 312 and 1250 Hz) at each of four signal levels (10, 30, 50, and 70 dB nHL). The averaged waveforms then were subjected to various analyses including bandpass filtering (using digital filters centered at the test frequency and its second harmonic), spectral analyses for selected post-stimulus intervals and cross correlation (with the input stimulus). Preliminary review of the data provides evidence of the presence of the FFP at signal frequencies yielding the largest response and of marked alterations in response complexity as a function of stimulus frequency and level. Results of more detailed analyses and comparison of the FFP with responses to clicks and click trains will be presented. [Work supported by NINCDS.]

THURSDAY MORNING, 10 NOVEMBER 1983

GARDEN ROOM, 8:30 TO 11:52 A.M.

Session II. Speech Communication IV: Speech Perception

Ian Maddieson, Chairman

Phonetics Laboratory, Linguistics Department, University of California, 405 Hilgard Avenue, Los Angeles, California 90024

Chairman's Introduction—8:30

Contributed Papers

8:35

III. Identification of transition frequency and duration changes in synthetic speech as a function of hearing status and age. Peter J. Ivory and John F. Brandt (Bureau of Child Research Laboratories, University of Kansas, Lawrence, KS 66045)

The identification of synthetic speech syllables which incorporated identical formant transition frequency changes in three different transition duration contexts (40, 120, 240 ms) or which incorporated identical transition duration changes in three different transition frequency contexts (where F_2 locus was 800, 1520, or 2120 Hz) was assessed for young

and old normal-hearing and sensorineurally hearing-impaired listeners. The older listeners, independent of hearing-impairment, demonstrated a significant decrement in identification performance for all experimental continua. The hearing-impaired listeners, independent of age, demonstrated a significant decrement in identification performance for most, but not all, experimental continua. Specifically, identification of syllables incorporating frequency changes was not significantly affected in the 240-ms duration context but significantly poorer in the contexts utilizing the two shorter transition durations. Furthermore, identification of syllables incorporating duration changes was not significantly affected in the 800-Hz transition locus context but significantly poorer in the higher frequency contexts.

II2. Speech understanding of older adults: A preliminary study. Lois L. Elliott, Laura Lyons, and Lu Ann Busse (Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

Forty-two normal older adults, aged 60–75 years, were tested on a battery that included the Speech Perception in Noise (SPIN) test, conventional, and experimental pure tone sensitivity, the W-22 test in quiet and noise, open- and closed-set identification of synthesized syllables that differed in the place of articulation feature, syllable discrimination, the Block Design subtest of the WAIS and the Concept Formation subtest of the Woodcock–Johnson Psychoeducational Battery. Results showed considerable variability on all measures. Only performance on the W-22 Test *in noise* and sensitivity at 4 kHz were significantly related to absolute levels of performance on low- (LP) and high-predictability (HP) SPIN test sentences. No test variables were significantly related to *relative* performance on HP and LP sentences. The differing patterns of results for different subjects of this homogeneous sample having excellent language skills suggested that: (1) a larger study in which linguistic abilities were systematically manipulated might produce a different outcome and (2) data for 400–500 similar older adults might show significant but differing *patterns* of processing auditory stimuli, particularly in the extent to which cognitive processes contribute to listening. [Work supported, in part, by NSF.]

8:59

II3. Exploring the “McGurk effect.” Sharon Y. Manuel, Bruno H. Repp, Michael Studdert-Kennedy, and Alvin M. Liberman (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

McGurk discovered that when a videotape of a talker is dubbed with similar but mismatched utterances, the optic information exerts a strong influence on the speech that is perceived, often without the observer's awareness. In several studies, we explored further the effects of conflicting acoustic and optic information on consonant perception. The following questions were asked: (1) Do subjects report a consonant even when no consonantal cues are contained in the acoustic signal (an isolated vowel)? (2) How is acoustic–optic place of articulation conflict resolved when acoustically specified manner (e.g., nasal, stop) restricts possible places of articulation? (3) How far does observers' awareness of acoustic–optic discrepancy reduce cross-modal integration? [Work supported by NICHD.]

9:11

II4. Judging sine wave stimuli as speech and as nonspeech. David R. Williams, Robert R. Verbrugge, and Michael Studdert-Kennedy (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Interest in the perceptual equivalence of “static” and “dynamic” stimuli spans both speech and nonspeech domains. Previous investigations of speech stimuli have shown that, given matched sets of three-formant steady-state vowels and vowels flanked by consonantal transitions, phonetic equivalence is not defined by equal position along the two continua [B. Lindblom and M. Studdert-Kennedy, *J. Acoust. Soc. Am.* **42**, 830–843 (1967)]. For judgments of relative pitch, on the other hand, ramp tones are judged to be nearly equivalent in pitch to steady-state tones at their endpoint frequencies [P. T. Brady *et al.*; *J. Acoust. Soc. Am.* **33**, 1357–1362 (1961)]. The present set of experiments demonstrates this difference in categorization of static and dynamic stimuli using a single type of stimulus. Two sets of sine wave stimuli (FLAT/CONTOUR) were constructed by modeling the frequency and relative amplitude characteristics of the aforementioned speech stimuli. When judged in terms of phonetic categories (/u/ vs /i/), significantly greater proportion of /u/ judgments were observed for the FLAT than for the CONTOUR continuum. However, when the same two sets of stimuli were judged in terms of relative pitch (HIGH versus LOW), no significant difference in response pattern for the two continua was noted. Thus the experiments demonstrate that two distinct response patterns may be elicited for sine wave stimuli contingent only on the experimenter's instructions. [Work supported by NICHD.]

II5. Perception of sine-wave analogs to individual formants of CV syllables. Rosemary Szczesiel, Richard E. Pastore, and Lawrence Rosenblum (Department of Psychology, State University of New York at Binghamton, Binghamton, NY 13901)

The immediate goal of our research was to delineate the nature of the perceptual consequences of the types of acoustic changes typical of phoneme contrasts. In this study we used sine-wave analogs of individual formant components of CV syllables. All of our sine-wave stimuli were dynamic in that they were characterized by various changes in frequency as a function of time which mimicked characteristic formant transitions. Native listeners performed both labeling and discrimination tasks in response to individual formant equivalents. Preliminary results suggest some similarity between the perception of speech and sine-wave analogs to formant components both in the demonstration of categorical perception and the location of category boundaries. In general, the relatively distinct category boundaries seldom corresponded to those which would be predicted by simple inspection of visual representations of the physical stimulus characteristics. [Work supported by NSF grant BNS 8003704 to the second author.]

9:35

II6. Perception of “modulation” in speechlike signals II: Discrimination of frequency extent as a function of spectral content and direction of change. Creighton J. Miller and Robert J. Porter, Jr.¹ (Kresge Hearing Research Laboratory of the South, LSU Medical Center, 1100 Florida Avenue, Bldg. 124, New Orleans, LA 70119)

This is the second in a series of studies designed to evaluate listeners' perception of modulation in speechlike signals [R. Porter and C. Miller, *J. Acoust. Soc. Am. Suppl.* **1** 73, S3 (1983)]. The study examines discrimination of second formant stimuli whose center frequency traversed a single 120 ms haversine modulation at their midpoint. Extent of frequency change for the stimulus set was varied in 20-Hz steps across the range, 100–400 Hz, both above and below the steady-state value of 1232 Hz. Separate continua were generated with and without the presence of a 769-Hz first formant. Paired comparison discrimination tests were conducted with listeners trained to judge which of the stimuli changed most. Results will be discussed in terms of their relation to the modulation percept, and to prior results of other investigations which have studied the perception of second formant transitions in and out of speech contexts. [Supported in part, by NINCDS NS #11647, The Louisiana Eye and Ear Foundation, The Louisiana Lions, and The Kresge Foundation.]¹ Also Department of Psychology, University of New Orleans.

9:47

II7. Identification and discrimination of visually presented synthetic consonant–vowel stimuli. Brian E. Walden and Allen A. Montgomery (Army Audiology and Speech Center, Walter Reed Army Medical Center, Washington, DC 20307)

Two-dimensional vector-based dynamic images representing the consonant–vowel syllables /ba/, /va/, and /wa/ were produced on an intelligent graphic system. The starting configuration and steady-state vowel were identical for these three stimuli. Visual presentation to subjects revealed these stimuli to be highly identifiable representations of the three consonants. Six intermediate stimuli were also generated between each of the three possible pairs of consonants and represented equal transitional steps between the exemplars. The three resulting eight-item continua were presented for identification and discrimination (ABX paradigm). The data revealed that the slope of the identification function in the region of the phoneme boundary and the location of the 50% point differed for the three continua. Identification and discrimination functions, as well as reproductions of stimulus frames, will be presented to illustrate the phoneme boundaries and most distinctive phonemic cues.

9:59

II8. Comparison of three performance measures obtained with perceptually balanced vowel continua. B. Espinoza-Varas (Psychology Department, University of Calgary, Calgary, Alberta T2N 1N4, Canada)

The auditory representation of formant frequency appears to be consistent with a scale of critical-band units: the bark scale [Schroeder *et al.*, *J. Acoust. Soc. Am.* **66**, 1647–1652 (1979)]. This paper compares measures of identification, discrimination, and perceptual distance obtained in vowel continua divided into “perceptually equal” steps; i.e., equal steps along the bark scale. The continua ranged from either /u/ to /i/ or from /ɔ/ to /ɛ/. The 400-ms, steady-state vowels were generated using a three-formant parallel synthesizer. Within each continua, the frequencies of F_1 and F_3 were fixed, while the frequency difference $F_2 - F_1$ (or $F_3 - F_2$) varied from ≈ 0.5 to ≈ 7.0 bark in steps of ≈ 0.5 bark. The fundamental frequency was 120 Hz. Listeners were asked to identify each of the stimuli in the continua and to make a “vowel-goodness” rating. The results are discussed in the context of the question of whether the bark scale is consistent with all three performance measures [Supported by AHFMR and NSERC.]

10:11

II9. Effect of voice type and message format on speed and accuracy of response to voice warnings. Timothy R. Anderson (Air Force Aerospace

Medical Research Laboratory, Wright-Patterson AFB, OH 45433), Jay Freedman, and Andy Rumbaugh (Air Force Institute of Technology, Wright-Patterson AFB, OH 45433)

The effect of LPC voice type and of message format on the speed and accuracy of response to voice warning signals were investigated at various levels of a simulated military inflight noise environment. LPC voice types were male, female, and machine, and the message formats were voice warning, repeat voice warning, and tone-voice warning. The primary task was a manual tracking task that required continuous attention whereas the secondary task required individual manual responses to randomly presented warning messages. A background signal of inflight aircraft voice communications was presented via the subject's earphones throughout the experiment. The voice warning messages were also presented via the earphones at levels of 0, 5, and 10 dB above this background communication level. Ten subjects completed the experimental tasks while wearing standard Air Force flight helmets and oxygen masks in the presence of the emulated aircraft noise at levels of 105 and 115 dB (SPL). The effects of LPC voice type and message format on voice warning effectiveness will be presented and discussed.

10:23–10:40

Break

Contributed Papers

10:40

III10. Sensitivity to rate-of-change of frequency transition. Donald G. Jamieson and E. Slawinska (Department of Psychology, University of Calgary, Calgary, Alberta T2N 1N4, Canada)

We examined a possible psychoacoustic basis for the relatively invariant duration over which the rapid formant frequency transitions cueing place of articulation occur in initial-position stop consonants such as /ba/. Our stimuli consisted of a 200-ms duration compound of a pure-tone frequency glide followed by a steady-state pure-tone stimulus. Glides covered a fixed range of frequencies (400–700 Hz; 1500–2200 Hz; or both together) over a variable interval of time (10–90 ms, in 10-ms steps). Three experiments examined discrimination performance in an AX paradigm; on any trial $X = A$ or $X = (A \pm \text{one step in a stimulus series})$. In each experiment, discriminability ($-\ln \mu$) tended to achieve a maximum, and bias ($\ln \beta$) tended to achieve a minimum, when the transition occurred over 50 ms. These results support and extend earlier reports [A. Nabelek and I. Hirsch, *J. Acoust. Soc. Am.* **45**, 1510–1519 (1969)] and appear to reflect a special perceptual sensitivity to frequency changes which occur within the 40–60-ms interval of time typical for the analogous transitions in natural speech. [Work supported by NSERC and AHFMR.]

10:52

III11. Fundamental frequency and the comprehension of simple and complex sentences. Leah S. Larkey and Martha Danly (MIT 36-511, Speech Communication Group, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

The intelligibility of sentences with monotone fundamental frequency was assessed using a sentence verification paradigm, which measured listeners' reaction time in judging whether a sentence was true or false. A zero-phase vocoder was used to allow manipulation of fundamental frequency contours and resynthesis resulting in speech that sounded as much like natural speech as possible. Sentences were vocoded and resynthesized using either a monotone contour or the original contour. Listeners' reaction times to monotone short, simple sentences were slower by 48 ms than their reaction times to the same sentences with normal fundamental frequency contours. Reaction times to longer sentences with complex syntactic structures will also be presented. The results suggest that an adequate model of speech perception and comprehension must incorporate intonation. The contributions of segmental effects, terminal fall, stressed syllables, and declination are considered. [Work supported by NIH and the Sloan Foundation.]

11:04

III12. Precategorical Acoustic Store (PAS): Is there an auditory sensory memory? Diane Williams (Psychology Service (116-B), VA Medical Center, 150 South Huntington Avenue, Boston, MA 02130)

The existence of an auditory sensory memory called Precategorical Acoustic Store (PAS) was proposed by Crowder and Morton [*Percept. Psychophys.* **5**, 365–373 (1969)] to account for both the modality and suffix effects obtained in list-learning experiments. While PAS provides a good explanation of these effects, the memory has little independent empirical support. PAS was directly investigated in two experiments. In the first experiment, the subject heard a subliminally presented word. The subject's task was to determine whether the two words were the same. In the second experiment, the subjects heard four words presented simultaneously at normal intensity, and later read a word. The subjects' task was to determine whether the word he or she read was one of the words that were heard. If the auditory stimulus was represented in PAS, then performance should be better if the visually presented probe appears before PAS has appreciably decayed, and performance should decrease with increasing probe lags. In both experiments, discriminability performance was roughly constant with increasing probe lags. This failure to support the proposed PAS is discussed.

11:16

III13. The pursuit of invariance in speech signals. Leigh Lisker (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510, and University of Pennsylvania, Philadelphia, PA 19104)

We may imagine a modified Muller-Lyer illusion in which two lines of unequal length are perceived as equal in different contexts. Is this analogous to the common situation in speech, where two distinct signals, e.g., *deep* and *duke*, are judged “equal” with respect to their /d/'s? In the visual case the term “illusion” is susceptible of precise definition, allowing us to speak of a mismatch between stimulus and percept. In the speech situation we are confronted with problems that make it very difficult to pose, let alone answer the question as to whether the /d/ common to *deep* and *duke* reflects a perceptual invariance that is illusory. Identity of spelling is only weak evidence for perceptual constancy, since spelling is more akin to phonological than phonetic representation, and at least some phonemes comprise phonetically distinct “sounds.” An example is provided by the voiceless stops of English. It can be shown that acoustically identical segments contribute to *bdg* in some contexts and to *ptk* in others.

Another example involves /wy/, which may be produced by the same vocal tract configuration in different contexts. Even if some phonemes are marked by acoustic invariants, not all of them are. [Work supported by NICHD.]

11:40

III5. Neural response patterns to speech sounds—A model. Pierre L. Divenyi (Speech and Hearing Research Facility, V. A. Medical Center, Martinez, CA 94553 and I.N.R.S.—Telecommunications, Université de Québec, Verdun, Quebec H3E 1H6, Canada), Robert V. Shannon (Department of Otolaryngology, University of California, San Francisco, CA 94143), and Stephen R. Saunders (Bell-Northern Research, Verdun, Quebec, H3E 1H6, Canada)

11:28

III4. Exploiting lawful variability in the signal: The TRACE model of speech perception. Jeffrey L. Elman and James L. McClelland (Department of Linguistics and Psychology, University of California, San Diego, La Jolla, CA 92093)

We describe a model of speech perception in which lawful variability in the speech signal is treated as a source of additional information, rather than as noise. Excitatory and inhibitory interactions among nodes for phonetic features, phonemes, and words are used to account for aspects of coarticulation, as well as the interaction of bottom-up and top-down processes in perception of speech. Two results from a working computer simulation of this model are presented. First, we show how the simulation is able to process real (digitized) speech and retune the detection of features and phonemes in accord with the context. Second, we demonstrate how perceptual behavior which appears to be guided by rules can be induced without any explicit rules in the system. [Work supported by the Office of Naval Research.]

We propose a four-stage model to represent auditory response magnitude over time and across frequency channels. The first stage performs spectral estimation by computing power at each of the output frequency channels. Frequency is scaled in basilar membrane distance and the input is passed through a window having a duration inversely proportional to the frequency. The next stage has been described earlier [R. V. Shannon, *J. Acoust. Soc. Am. Suppl.* 1 65, S56 (1979)]; it portrays frequency analysis in the cochlea. This stage includes mechanisms of cochlear filtering (by means of two filter banks, one having sharp and the other having broad filters), nonlinear compression of the input power, and lateral suppression. The third stage models temporal adaptation in the auditory nerve. The final stage is a temporal integrator. Speech sounds analyzed by the model acquire several interesting characteristics: (i) The dynamic range of the input is greatly reduced (< 15 dB), (ii) bandpass information (e.g., the one in formants, fricatives, etc.) is represented as a spectral *edge* at the low side of the bandpass region, (iii) bursts (especially high-frequency plosives) acquire temporal sharpness, (iv) individual glottal pulses are clearly visible only at high frequencies. [Work supported by Institut National de la Recherche Scientifique, the Veterans Administration, and grants by N.I.H.]

THURSDAY MORNING, 10 NOVEMBER 1983

CABINET ROOM, 9:00 A.M. TO 12:20 P.M.

Session JJ. Shock and Vibration III: Statistical Methods for the Solution of Practical Vibration and Acoustical Transmission Problems

James A. Moore, Chairman
Cambridge Collaborative, 225 Third Avenue, Cambridge, Massachusetts 02142

Chairman's Introduction—9:00

Invited Papers

9:05

JJ1. Phase estimation for transfer function of multi degree-of-freedom-systems. Richard H. Lyon (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

The use of statistical estimates for vibratory response is now a well-established procedure using the methods of SEA. It has generally been assumed that while one could estimate the average response in frequency bands, statistical methods can say little about the phase of response. But the phase is very important in some applications of transfer function analysis, such as waveform reconstruction, and it would be very desirable to have procedures for estimating transfer function phase on the basis of general structural features in the way that SEA uses such features for estimating average response. Some recent developments in the estimation of the phase of structural transfer functions show promise that phase can be estimated from general system parameters. The use of phase information in waveform reconstruction and the role of these new developments in designing inverse filters for such reconstructions are reviewed.

9:35

JJ2. An approach to the statistical energy analysis of complex systems. Richard G. DeJong (Cambridge Collaborative, Inc., P. O. Box 74, Cambridge, MA 02142)

The principles of SEA have been used to formulate a procedure to analyze the dynamic response of complex systems. This procedure allows for individual components of a system to be modeled separately and then combined with other components to construct a complete system model. An analysis is presented that shows

that the requirement for a high modal density in each component can be relaxed under certain conditions which allows the procedure to be used at lower frequencies. The procedure has been implemented in a computer program, and some examples of its application and comparisons to measurements are given.

10:05

JJ3. Statistical energy analysis for beam networks in building structures. M. J. Sablik, R. E. Beissner, H. S. Silvus, Jr., and M. L. Miller¹ (Southwest Research Institute, P. O. Drawer 28510, San Antonio, TX 78284)

The statistical energy analysis method is applied to a beam network in a building structure. A method of taking structural resonances into account is introduced into the analysis. Coupling loss factors are computed using transmission coefficients presented in earlier work [M. J. Sablik, *J. Acoust. Soc. Am.* **72**, 1285 (1982); L. Cremer, M. Heckl, and E. E. Ungar, *Structural Borne Sound* (Springer-Verlag, New York, 1973)]. Transfer functions computed from the model beam network are compared over a wide frequency range to experimentally measured transfer functions from an equivalent beam network in an existing building structure in which one of the beams is vibrationally excited. Our results emphasize that unless resonance effects are taken into account, the predictions of statistical energy analysis will not display the fine details found in the frequency dependence of transfer functions for real building structures. ¹ Formerly with SwRI.

10:35

JJ4. Helicopter airframe vibration transmission modeling using statistical energy analysis (SEA). Charles A. Yoerkie (Sikorsky Aircraft Division, United Technologies Corporation, Stratford, CT 06602)

Predicting helicopter cabin noise is a complex issue. It involves sources which are both acoustic/aerodynamic and vibratory in nature as well as a compact, highly interconnected airframe structure. Energy transmits to the cabin via both airframe structureborne and airborne paths. Since a large number of resonant structural and acoustic modes are excited, a statistical approach, which treats the response of the modes collectively in groups, is desirable. The method described for modeling the helicopter airframe and cabin interior is Statistical Energy Analysis (SEA). An SEA model, funded by a NASA Langley Research Center contract, is presented for the Sikorsky S-76 (a 4-13 passenger commercial helicopter). This overall source-to-receiver modeling approach allows for evaluation of various noise control measures in the conceptual design stage. Expressions have been developed for SEA parameters and the coupling loss factor has been related to the more familiar vibration transmission coefficient. A full scale measurement program was conducted on the Sikorsky S-76. Data are presented showing the SEA parameters and comparisons with the analytic model. These data include vibration transmission along the beam/panel structure and transmission through intermediate acoustic spaces connected by both resonant and nonresonant conditions. In addition, data are presented concerning the effects of coherent and incoherent source excitations at distant response points. [Work supported by NASA Langley.]

11:05

JJ5. A comparison of measured and predicted sound levels in the space shuttle payload bay. John F. Wilby, Allan G. Piersol, and Emma G. Wilby (Bolt Beranek and Newman Inc., P. O. Box 633, Canoga Park, CA 91305)

An analytical model to predict space-average sound levels in the payload bay of the Space Shuttle at lift-off has been developed using power balance methods. The development of the model has been reported earlier [L. D. Pope and J. F. Wilby, *J. Acoust. Soc. Am.* **62**, 906-911 (1977)]; it is now possible to compare the predictions of the model with data from the first three launches. Space-average sound levels in the bay were estimated from measured data and the resulting spectra compared with predictions; discrepancies were observed in the frequency range above 125 Hz. Modifications made to the analytical representation for the payload bay door resulted in much better agreement between measured and predicted sound levels. Payloads present in the bay were modeled analytically, but the predicted effects on space-average sound pressure levels were small because of the small size of the payloads. [Work performed under NASA Contract NAS5-26570.]

Contributed Papers

11:35

JJ6. Power transmission across a complex plate branch. Robert C. Haberman, Donald T. Pan, and Thomas S. Graham (Bolt Beranek and Newman Inc., Union Station, New London, CT 06320)

Finite element and classical techniques were combined to compute the transmission and reflection of vibratory energy across a plate branch with structural connections too complex to be modeled with classical techniques alone. The procedure for combining the models is outlined and numerical results are presented. Results are given for a simple plate connection (which agrees with classical theory) and several complex welded and riveted plate connections. For complex connections, results are shown to be very sensitive to structural details such as mass and contact area at the branch. Since contact area is in general unknown, this suggests that power transfer coefficients should be averaged over frequency bands.

The method described may be used to compute coupling coefficients for SEA models.

11:50

JJ7. Point-force random excitation of a thin plate bent into an L-shape. Emiliios K. Dimitriadis and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The authors' model for random point-force excitation [*J. Acoust. Soc. Am. Suppl.* **1** **73**, S71 (1983)] is refined and simplified. The modal cross-correlation terms introduced by point forces are accommodated by using the method of images with only the nearest edge taken into account to

approximate the power input to a finite plate. Arguments analogous to those used in room acoustics [see for example, *Acoustics* (McGraw-Hill, 1981), p. 295] allow the average cross-correlation term to be expressed in terms of the power input ratio, finite plate to infinite plate. Statistical energy analysis (SEA) parameters are computed for the L-configuration plate for the cases when all edges are simply supported and when all edges are free, with Warburton's Rayleigh-Ritz modes used in the latter case. The analysis confirms that the SEA parameters are very nearly independent of force position. Calculations compare favorably with experiments conducted on a freely suspended L-configuration plate with different source locations and two different damping configurations. [Work supported by Whirlpool Corporation.]

12:05

JJ8. Application of SEA in developing a high-frequency vibration test method. Jerome E. Manning (Cambridge Collaborative, Inc., P. O. Box 74, Cambridge, MA 02142)

The vibration environment in many aerospace environments extends to very high frequencies (~100 kHz). However, because of the limitation in test equipment, it is difficult and often impossible to impose a vibration specification on an equipment supplier that extends beyond 4000 Hz. This presentation illustrates the use of SEA to develop the requirements for a new test technique.

THURSDAY MORNING, 10 NOVEMBER 1983

GOLDEN WEST ROOM, 8:30 TO 11:55 A. M.

Session KK. Psychological Acoustics IV: Discrimination of Frequency and Periodicity

Peter M. Narins, Chairman

Department of Biology, University of California, Los Angeles, California 90024

Chairman's Introduction—8:30

Contributed Papers

8:35

KK1. Effect of duration on pitch discrimination of complex signals Virginia M. Richards and Ervin R. Hafter (Department of Psychology, University of California, Berkeley, CA 94720)

Discrimination of modulation frequency was measured using trains of n clicks ($n = 4$ to 52) presented at a rate of 250/s. Each click was the product of a 3500-Hz sinusoid and a Gaussian envelope. Three spectral envelopes were tested, having standard deviations of 350, 560, and 933 Hz. The rate of improvement in performance over duration is reflected in a plot of log-threshold vs log- n . These functions are well defined by two straight lines: for small values of n , the slopes are near -1.0 , and for larger n 's, the slopes approach -0.5 . This suggests a model in which two factors influence the discrimination process. For small n 's, the spectral width of individual components in the amplitude spectrum are reduced in proportion to the length of the click train; thus the slope of -1.0 . With further increases in n , statistical averaging reduces the effect of neural noise by a factor of the square-root of n ; hence the slope of -0.5 . The model fit well to all of the data, except those obtained with the widest spectral envelope. This may be due to greater sensitivity to temporal information available in the low-frequency regions of these wideband clicks. [Supported by NIH.]

8:55

KK2. Asymmetrical frequency discrimination in primates. Joan M. Sinnott and Michael R. Petersen (Psychology Department, Indiana University, Bloomington, IN 47405)

Auditory frequency difference limens at 0.5, 1.0, 2.0, and 4.0 kHz were measured monaurally in humans and in four Old World monkey species: Japanese and rhesus macaques (*Macaca fuscata*, *Macaca mulatta*) and vervet and deBrazza monkeys (*Cercopithecus aethiops*, *Cercopithecus neglectus*). A go, no-go, repetitive-standard discrimination procedure was employed with positive reinforcement. At certain frequencies and sensation levels, humans and macaques exhibit superior discrimination for frequency increments, while under these same conditions the *Cercopithecus* species exhibit superior discrimination for frequency decrements. Possible coding mechanisms to account for these results will be discussed, including monitoring rate increases versus rate decreases in asymmetrically

tuned auditory nerve fibers. In one species, the Japanese macaque, there is some evidence that superior discrimination of frequency increments may relate to the design features of the animal's vocal communication system. [Supported by NSF, Sloan Foundation, and Deafness Research Foundation.]

9:15

KK3. Temporal microstructural rules for periodicity of random-amplitude sequences. Irwin Pollack (Mental Health Research Institute, University of Michigan, Ann Arbor, MI 48109)

We examine microstructural rules for the emergence of periodicity formed by temporal subpatterns within repeated random-amplitude sequences. The specific subpatterns were weighted either by successive replication, e.g., AAABAAAB ... or by differential amplitude weighting, e.g., AAbAAb ..., where b is a reduced version of A. For short-duration subpatterns, the dominant periodicity is that of the entire repeated sequence. For long-duration subpatterns, the dominant periodicity is that of the individual subpatterns, depending upon the relative amplitude weighting and the number of replications. [Work supported by NSF.]

9:35

KK4. Detection of temporal changes in the pitch of complex stimuli. William A. Yost and Morris Moore (Parly Hearing Institute, 6525 N. Sheridan Rd., Loyola University, Chicago, IL 60626)

The repetition pitch of ripple noise was temporally varied to determine the relationship between the rate of pitch variation and the perception of pitch change. Listeners were asked to discriminate between flat spectral noise and ripple noise whose ripple density was temporally varied in a sinusoidal manner. The spectrum of the temporally varying ripple noise stimulus was $1 + m \cos[2f\{g(T)\}]$; where f is frequency, m is the modulation depth of the spectral ripple, and $g(T) = T + T \sin 2ft$, where T is the delay used to generate ripple noise, T is the range of pitch change, f is the rate (Hz) of pitch change, and t is time. The repetition pitch of ripple noise is directly related to the delay, T . At each rate of pitch variation the depth of modulation (m) was varied in the adaptive, forced-choice task until the listener was at 71% correct in his or her ability to discriminate ripple noise

from flat noise. For a variety of conditions the results indicate that listeners are unable to detect the change in the pitch of ripple noise at rates of pitch change that exceed 2–5 Hz. This in turn implies that the auditory system is slow (200–500 ms estimated integration time) in its ability to process the complex stimulus of ripple noise. [Work was supported by the National Science Foundation.]

9:55

KK5. Rate discrimination of high-pass filtered pulse trains. Glenis R. Long and John K. Cullen, Jr. (Department of Otorhinolaryngology, Louisiana State University Medical Center, New Orleans, LA 70119)

Difference limens (DLs) for trains of 30- μ s impulses were determined for repetition rates of 50-, 100-, 200-, 400-, and 800 pulses per second under conditions of no filtering and high-pass filtering (115 dB/oct) with corner frequencies of 2.5-, 5.0-, 7.5-, and 10 kHz. Noise, at 15-dB spectrum level, was low passed (115 dB/oct) at the corner frequency of each high-pass filter condition and mixed with the trains of pulses to preclude discrimination on the basis of potential low-frequency signal compounds. Measures were obtained from four trained listeners at a signal level of 30 dB SL relative to individually determined thresholds for each filter condition and repetition rate. A two alternative forced choice adaptive procedure was used. The data support the hypothesis that resolution of pulse-train repetition rate involves both temporal and pitch-based processes; the latter becoming ineffective when frequency resolution of the ear is insufficient to resolve separate harmonics of the signal. [Work supported by NIH.]

10:15

KK6. Temporal tuning curves and their relation to frequency tuning curves. Anna C. Schroder and Edward Cudahy¹ (Syracuse University, 805 South Crouse Avenue, Syracuse, NY 13210)

Temporal tuning curves for a 1000-Hz, 10-ms, 10-dB SL sinusoidal signal with a 500-, 1000-, or 1500-Hz, 10-ms sinusoidal masker were measured using a 2IFC adaptive procedure. Frequency tuning curves for the 10 dB SL, 1000-Hz signal were also measured for 10- and 500-ms simultaneous maskers. Five normal hearing listeners were employed. The frequency tuning curves for the brief masker were broader than those for the long masker, especially on the high-frequency side of the signal. Marked differences were not found between the temporal tuning curves for the three masker conditions. The implications of these results for relations between auditory frequency and temporal analysis will be discussed. ¹ Present address: House Ear Institute, 256 South Lake Street, Los Angeles, CA 90057.

10:35

KK7. Generalization of learning in auditory pattern discrimination. M. R. Leek (Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85287)

Frequency discrimination for tones embedded in a varying sequence of ten brief tones can be extremely poor. Yet after repeated exposure to a constant tonal sequence, listeners demonstrate discrimination of changes in a target component almost as fine as for the target tone presented alone [e.g., C. S. Watson *et al.*, *J. Acoust. Soc. Am.* **60**, 1176–1186 (1976)]. It is clear, however, which aspects of the constant pattern are learned, thereby allowing the listener to overcome the effects of the surrounding tonal context on discrimination of the target. This experiment investigated generalization to new patterns of the learned target and learned context to assess their contributions to pattern discrimination. Four listeners were trained in a same–different task to discriminate small frequency changes in a 920-Hz target tone, which was the second element in a ten-tone pattern. Frequency difference limens were subsequently determined for the learned target tone embedded in a different tonal context, and for different targets when the context was held constant. Listeners were unable to maintain small difference limens for the target tone when the surrounding elements of the pattern were changed. Varying the target in the learned context, however, proved much less disruptive. Variations in the frequencies and the frequency contour of the learned context served to illustrate

the relative importance of these characteristics. Results indicated that direct experience with a target tone contributes little to discrimination performance as long as the frequency contour of the pattern remains stable within certain limits.

10:55

KK8. Detection of tones in synthetic plosive bursts. C. L. Farrar, Y. Ito, P. M. Zurek, N. I. Durlach, and C. M. Reed (Research Laboratory of Electronics, 36-767, MIT, Cambridge, MA 02139)

The general goal of our research is to understand discrimination between sounds that differ only in spectral shape. The objective of this specific study was to test the notion that the discriminability of two sounds is monotonically related to the difference between their simultaneous masking pattern (or, its corollary, that two sounds that produce equal masking are indiscriminable.) Measurements were made of the thresholds for tones in two masking noises, one with a /p/-shaped spectrum and the other with a /t/-shaped spectrum. Data on discriminating between these two masking noises have been previously obtained. We attempt to relate discrimination and detection results through transformations of “masking difference patterns,” which show the difference, as a function of frequency, between signal thresholds for the two maskers. Further, we will describe our progress in developing a general psychoacoustic model for discriminating between sounds that differ only in their spectral shapes.

11:15

KK9. Development of auditory thresholds in young chickens. Lincoln Gray (Department of Otolaryngology—Head and Neck Surgery, University of Texas Medical School, Houston, TX 77030) and Edwin W. Rubel (Department of Otolaryngology—Head and Neck Surgery, University of Virginia Medical School, Charlottesville, VA 22908)

Absolute auditory thresholds were estimated in chickens within 12 h of birth and at 4 days after hatching. A two-interval, forced-choice staircase procedure was modified to use momentary suppressions in chick's peeping as indications of correct and incorrect responses. The estimated thresholds of both ages were the same at low frequencies (250–500 Hz). At higher frequencies (1–2 kHz), however, 4-day-old chicks had thresholds about 20 dB lower than the 0-day-old chicks. Thus absolute thresholds are a function of both frequency and age. These thresholds are likely to reflect perceptual development for two reasons: (a) Similar thresholds from both ages at low frequencies show that developmental differences are not due to differences in the sensitivity of the testing procedure; and (b) thresholds obtained from the 4-day-old birds are similar to other estimates from mature birds. In conclusion, responsiveness to low frequencies develops before responsiveness to higher frequencies. [Work supported by NIH.]

11:35

KK10. Phase effects and profile analysis. David M. Green and Christine R. Mason (Laboratory of Psychophysics, Harvard University, 33 Kirkland Street, Cambridge, MA 02138)

Randomizing the phase of the components of a tonal complex on each presentation has little effect on detection of an intensity increment of the central component. The complex consisted of eleven equal amplitude tones ranging from 200 to 5000 Hz. The spacing between adjacent components was equal on a logarithmic frequency scale. The signal was a sinusoid added in-phase to the central, 1000-Hz component. The overall sound pressure level of the sound was varied on each presentation according to a rectangular distribution with a range of 40 dB and a median sound pressure level per component of 45 dB. Over 50 complexes with different waveforms were generated by randomly selecting different starting phases for each component. Eight of these waveforms were selected, and each was presented as the complex for more than 500 trials in a 2AFC experiment. Thresholds for a signal added to each of these different complexes had a total range of only 2.4 dB. In a final test, we selected a different waveform on each presentation. Signal threshold was elevated by only 3 dB. [Work supported by NIH.]

Session LL. Psychological and Physiological Acoustics IV: Acoustics of Echolocating Animals Under Water

Whitlow W. L. Au, Co-Chairman
Code 512, Naval Ocean Systems Center, Kailua, Hawaii 96734

Ronald Schusterman, Co-Chairman
Biology Department, California State University, Hayward, California 94542

Chairman's Introduction—1:00

Invited Papers

1:05

LL1. Dolphin sound production: Physiologic, diurnal, and behavioral correlations. Sam H. Ridgway (Code 5142, Naval Ocean Systems Center, San Diego, CA 92152)

The mechanics of dolphin sound production will be reviewed but many questions remain. For example, do the diagonal membranes function somewhat as vocal cords? Does the nasal plug serve as a primary vibrating source? Do the nasofrontal sacs serve as tuned, or tunable, resonators? Do the nasal sacs serve as acoustic reflectors and help to focus sound? In general, there are three types of sounds produced by *T. truncatus* and other closely related dolphins—whistles, burst pulse sounds, and click trains. Click trains, generally of high peak frequency, are used by dolphins that are trained to perform echolocation tasks. All three types of sound have also been suggested as having potential value for communication. I have found that: (1) A single dolphin may produce as many as 50 000 separate sonic episodes in a single 24-h day (The average daily sound production is more like a tenth this figure); (2) Peaks in sound production occur in the hour before and during feeding, especially the first feeding of the day, and during the hour after sunset. It appears that dolphins display aggression and agitation by increasing the pulsatile content of sounds and by emphasizing the harmonic structure of whistles. It is possible that echolocation evolved from sounds first used in aggression.

1:35

LL2. Insights into dolphin sonar discrimination capabilities from broadband sonar discrimination experiments with human subjects. Whitlow W. L. Au and Douglas W. Martin (Naval Ocean Systems Center, P. O. Box 997, Kailua, HI 96734)

A wide variety of dolphin sonar discrimination experiments have been conducted, yet little is known as to the cues utilized by dolphins in making fine target discriminations. Hammer and Au [J. Acoust. Soc. Am. 68, 1285–1293 (1980)] found a correlation between dolphin discrimination capabilities and the matched filter response of the target echoes. Sonar discrimination experiments have also been conducted with human subjects using the same targets employed in the Hammer and Au and other dolphin experiments. When digital recordings made of the target echoes ensounded with a dolphinlike signal were played back at a slower rate to subjects, humans could make fine target discriminations under controlled laboratory conditions about as well as dolphins under less controlled conditions. Most of the cues reported by human subjects can be described by time-domain echo-features. The results of several dolphin sonar and human listening experiments will be discussed along with the target echo characteristics.

Contributed Papers

2:05

LL3. Critical ratio and bandwidth of the Atlantic bottlenose dolphin (*Tursiops truncatus*). Patrick W. B. Moore and Whitlow W. L. Au (Naval Ocean Systems Center, P. O. Box 997, Kailua, HI 96734)

Masked underwater pure-tone thresholds were obtained for an Atlantic bottlenose dolphin using an "up-down staircase" method of stimulus presentation and a "go/no-go" response procedure. Broadband noise at two levels was used to measure the animal's critical ratio at test frequencies of 30, 60, 90, 100, 110, 120, and 140 kHz. For frequencies of 100 kHz and below, the critical ratios were similar to those measured by Johnson [J. Acoust. Soc. Am. 44, 965–967 (1968)]. The results indicate a sharp increase in the critical ratio to 51 dB at 110 kHz followed by a slight decline to 46 dB at 120 kHz. Different bandwidths of noise with very sharp cutoffs were used to measure the dolphin's critical bandwidth at test frequencies of 30, 60, and 120 kHz. The critical bandwidth was approxi-

mately 10 times (10 dB) wider than the critical ratio at 30 kHz and 8 times wider (9 dB) at 60 kHz. At 120 kHz, the critical bandwidth was approximately the same as the critical ratio.

2:20

LL4. Peak sound pressure level and spectral frequency distributions in echolocation pulses of Atlantic bottlenose dolphins, *Tursiops truncatus*. Marion G. Ceruti, Patrick W. B. Moore, and Sue A. Patterson (Naval Ocean Systems Center, Hawaii Laboratory, P. O. Box 997, Kailua, HI 96734)

We investigated the relationship between peak sound pressure level (PPL) and the shape of the spectral frequency distribution of echolocation pulses emitted by Atlantic bottlenose dolphins (*Tursiops truncatus*) in a target detection task. Two dolphins were trained to station on a bite plate

and tail rest, echolocate, and report target condition (present or absent). A microprocessor data acquisition system [M. G. Ceruti and W. W. L. Au, *J. Acoust. Soc. Am.* 73, 1390-1392 (1983)], was used to collect amplitude and frequency data for each pulse and monitor subjects' responses. Ten sessions of 50 trials each (25 target present and 25 target absent, presented randomly) were chosen for each subject from sessions conducted during Nov. 1982. Pulses were classified as "bimodal" or "unimodal," depending on the number of maxima in the frequency spectrum between 30 and 135 kHz. For bimodal pulses, the degree of bimodality BI, also was measured by summing the maximum intensities using the intervening minimum as a baseline. The results collapsed across subjects yielded 15 670 bimodal and 12 383 unimodal pulses. The mean PPL for bimodal pulses was 207 dB re:1 μ Pa and for unimodal pulses, 197 dB. The Pearson product moment correlation coefficient between BI and PPL was 0.69. Peaks in the averaged bimodal spectrum occurred at 60 and 135 kHz or beyond, while the averaged unimodal spectrum was peaked at 120 kHz. These findings suggest that bimodal frequency distributions during high-pressure echolocation represents a physiological process in the pulse emission system.

2:35

LL5. Apparent echolocation by a sixty-day-old bottlenosed dolphin, *Tursiops truncatus*. Donald A. Carder and Sam H. Ridgway (Code 5142, Naval Ocean Systems Center, San Diego, CA 92152)

Little is known about the ontogeny of echolocation in dolphins. A bottlenosed dolphin calf born at our facility in San Diego Bay was observed continuously for two hours each day and randomly at other times. Squeals were heard about ten seconds after birth and whistle-like calls soon thereafter but we did not notice head scanning movements concurrent with high-frequency pulses until the dolphin's sixtieth day of life. On this day we noticed that the calf would examine items such as a human foot inserted in the water by approaching, head scanning, and pulsing. A hydrophone placed in the water elicited the same behavior. Seven pulse trains were recorded at distances of 1-3 m. Peak frequencies ranged from 33 to 120 kHz with 3-dB bandwidths of 28-81 kHz.

2:50

LL6. Target detection: Beluga whale and bottlenose dolphin echolocation abilities compared. Charles W. Turl and Ralph H. Penner (Naval Ocean Systems Center, Hawaii Laboratory, P. O. Box 997, Kailua, HI 96734)

An experiment compared the echolocation abilities of a beluga whale and a bottlenose dolphin to detect five targets of the same size and target strength (+/- 2 dB) at distances from 40 to 120 m. The performance for both animals at 40, 60, 80, and 100 m exceeded 90% correct detection; however, performance at 120 m ranged between 60% to 62% correct detection. The beluga whale and the bottlenose dolphin were tested with the same targets at 5-m increments at distances from 100 to 120 m. These test results indicated: (1) no significantly different performance ($p < 0.05$) between the beluga whale and bottlenose dolphin at the five test distances, (2) both animals differentially reported three of the five targets, independent of target distance, and (3) the performance of both animals at 120 m ranged between 60% to 90% correct detection depending on which target was presented at 120 m. There may be several reasons why both animals differentially reported three of the targets at ranges beyond 100 m, irrespective of absolute distance: (1) variable target strength, (2) the manner in which the targets were suspended plus rotational motion about the suspension point may have induced target echo variability, and (3) for targets under 100 m, target echo levels were large and both the beluga whale and bottlenose dolphin were able to differentiate the signal from the background noise. Beyond 100 m, subtle features of the targets determined their detectability. On two targets, the salient features were apparently unavailable at ranges beyond 100 m, and differential detection occurred. Because the range differences were small, the significant target feature that was absent was not solely a function of spherical spreading loss.

3:05

LL7. Bottlenose dolphin (*Tursiops truncatus*): Difference in the pattern of interpulse intervals. Ralph H. Penner and Charles W. Turl (Naval Ocean Systems Center, Hawaii Laboratory, P. O. Box 997, Kailua, HI 96734)

When the echolocation detection abilities of a beluga whale and a bottlenose dolphin were tested on identical cylindrical targets and distances, they had interpulse interval (IPI) distributions which were different, while detection accuracy showed no significant difference. Bottlenose dolphin pulse trains show a predictable and systematic relationship between the distance to a detected target and time between echolocation pulses, thus permitting an inference of attending distance [R. H. Penner and J. Kadane, *J. Acoust. Soc. Am. Suppl.* 1 68, S97 (1980)]. The beluga whale pulse trains contained, at all target distances, a first IPI component starting around 30 ms and increasing to 60 ms and a second IPI component between 200 to 220 ms which became more numerous as target distance increased. The transition zone between the first and second components contained very few interval counts. The interpulse intervals within the transition zone increased as target distance increased. If the bottlenose dolphin IPI patterns are accepted as models, and similar assumptions regarding attending distance are applied to the beluga whale pulse train, the picture is one of always scanning out to 30 m; followed by short scan out to 160 m. Subsequent target detection work using spheres instead of cylinders as targets, showed the same IPI distribution patterns for the beluga whale.

3:20

LL8. Comparative observations on odontocete sonar signals from captive animals. Cees Kamminga (Information Theory Group, Delft University of Technology, Mekelweg 4, Delft, The Netherlands)

In this paper a comparative analysis of typical sonar signals of several species of littoral and riverine dolphins in captivity is presented. The sonar wave shapes compared include those recently obtained from two species, whose acoustic behavior has not been described previously in literature, i.e., the off-shore *Lagenorhynchus albirostris* and the fresh water population of Irrawadi; *Orcaella brevirostris*. The latter fit very well in the picture established thus far from a large amount of echolocation data. A study of these sonar signals reveals the noteworthy fact that—apart from the Commerson's signal—they all share a rather simple, basic wave shape. This can best be understood and classified in terms of the well-known uncertainty relation in communication theory and enables us therefore to link bioacoustics with mathematical physics. The signals approach the theoretical lower bound for the product of the time duration and frequency bandwidth and could therefore be designated as "small time-duration bandwidth" signals. They are thus well suited as a class of elementary optimal time-frequency signals. The frequency range of interest in the first described class of cetacean sonar is situated at a dominant frequency below 100 kHz, while in the Phocaena and Commerson group this dominant value goes up to around 120 kHz. The latter thus enabling those small animals to resolve spatial differences in their habitat up to an order of magnitude of nearly a centimeter.

3:35

LL9. A qualitative model of the dolphin sonar. James W. Fitzgerald (Mystic Marinelife Aquarium, Mystic, CT 06355 and B-K Dynamics, Inc., New London, CT 06320)

By means of a complex, functional anatomy, the bioacoustic apparatus of the dolphin enables him to operate four separate sonar systems more or less simultaneously, using some of the anatomical components in common. These systems are: (1) a broadband high-frequency (~30-150 kHz) echolocation "attack" sonar, with a narrow (~10°) forward-looking beam pattern, that is used for the location and tracking of such targets as food-fish; (2) a broadband, low-frequency (< 30 kHz) echolocation "search" sonar, with a broad (~120°) forward-looking beam pattern, that is used for orientation and intruder detection; (3) an audio-frequency (~0.2-15 kHz), frequency-contoured "communication" sonar, with a substantially nondirectional beam pattern, that is used for communicating with other dolphins; and, finally, (4) a wideband (0.15-150 kHz), binaural "listening" sonar that is used as the sonar receiver for all three of the preceding active sonars, as well as a passive listening system for target detection, bearing, and classification. It is shown that echolocation "clicks" originating in the larynx, which enter the melon through the closed nasal plug, are totally internally reflected, radiating only from the forward end, and form the attack sonar. A logarithmic spiral of the form $R = A \exp\{-\tan \beta_0 \theta\}$,

where β_c is the critical angle, fits the shape of the melon. For the open nasal plug condition, echolocation "clicks" from the larynx propagate up the nasal duct into the premaxillary sac, which acts as a "bubble" transducer for the search sonar. The source of the whistle-like communicative phonations is identified as lip-modulations excited by exhaust air expelled from the premaxillary sac into the vestibular sac. Experimental confirmation, drawn in part from the bioacoustic literature on Odontocetes, is proffered in support of the principal qualitative features of this new model.

3:50

LL10. Echolocation in Harbour seals: Maybe they can do it after all. Deane Renouf (Department of Psychology/Marine Sciences Research Laboratory, Memorial University of Newfoundland, St. John's, Newfoundland, A1B 3X9, Canada)

Though there is indirect evidence to indicate that various species of seals are capable of some form of echolocation, all previous attempts at an experimental demonstration of sonar capacities have met with negative results. I have obtained the following circumstantial evidence that Harbour seals can echolocate: (1) During three breeding seasons I have observed apparently healthy blind adult harbour seals including females which have successfully raised pups. (2) When visual cues are reduced, harbour seals make click vocalizations similar to those of animals which are known to echolocate. (3) They are able to find live fish in total darkness, producing clicks while doing so. (4) One seal trained to retrieve a ring which he first had to find in a 10-m-diam tank performed as well in the dark as in daylight, but clicked only when he could not use vision. I obtained a more rigorous demonstration of Harbour seal sonar when I was able to train an animal to discriminate between two visually identical rings which differed in acoustic impedance. When the acoustical properties of the rings were made equal, the animal was no longer able to distinguish between them. Very faint single or doublet clicks were recorded during the animal's performance, and it is suggested that quiet distinct signals may have been appropriate in an enclosed tank when the objects to be discriminated were suspended close to the enclosure's walls. The reasons why the training procedures used in this demonstration were successful are discussed, and the limitations of the conclusions which can be drawn from such results are outlined.

4:05

LL11. Incidental evidence for echolocation in polar pinnipeds. Jeanette A. Thomas, Frank T. Awbrey, and Sheldon R. Fisher (Hubbs Sea World Research Institute, 1700 South Shore Road, San Diego, CA 92109)

Efforts to experimentally demonstrate echolocation in California sea lions, *Zalophus californianus* [W. Evans and R. Haugen, Bull. So. Cal. Acad. Sci. 62, 165-175 (1963); R. J. Schusterman, Psych. Rec. 16, 129-136 (1966)] and in grey seals, *Halichoerus grypus* [B. Scronce and S. Ridgway, in *Animal Sonar Systems*, pp. 991-993 (1980)] have been unsuccessful. However, similar studies on polar pinnipeds have not been conducted previously. No studies have investigated the potential for ultrasonic vocalizations in pinnipeds. Echolocation in polar pinnipeds has been suggested because of their highly developed vocal abilities and their need to find food and navigate during the dark austral winter [G. Kooyman, Ant. Res. Ser. 11, 227-261 (1968); J. Thomas and V. Kuechle, J. Acoust. Soc. Am. 72, 1730-1738 (1982)]. This presentation will summarize observations and evidence that indicates the presence of echolocation in polar pinnipeds and report the production of ultrasonic vocalizations by a captive leopard seal (*Hydrurga leptonyx*). Clicks, buzzes, and frequency-modulated chirps were produced with peak frequencies from 4 to 164 kHz, but generally between 50 and 60 kHz. This study demonstrates the importance of investigating ultrasonic vocalizations in all pinnipeds and implies that polar pinnipeds may be the best test group for echolocation.

4:20

LL12. Seal blindfolded discrimination: Echolocation not proven in *Halichoerus grypus*. B. L. Scronce and S. H. Ridgway (Code 514, Naval Ocean Systems Center, San Diego, CA 92152)

A four-year-old female gray seal from Iceland was trained to wear an opaque elastic band that blocked vision. Echolocation capability was evaluated in two experiments: (1) The seal was required to retrieve an air-filled plastic ring 20 cm in diameter placed at random in a 5 × 1 m section of a 10-m redwood tank. (2) The seal was required to detect the flat surface of a 25-cm styrofoam disk as opposed to the edge of the same disk (-30 dB target strength), placed randomly on either side of a divider. Without the blindfold, the seal's ring retrieval rate was 100% with a latency of 3.8 s. In 427 blindfolded trials there were 99% correct responses, but latency increased to 6.5 s. Head scanning movements were observed on about half the blindfolded trials and click trains were recorded on about 10% of the trials observed. In the second experiment, the blindfolded seal approached the divider and made a choice about target location (right or left) at least 1.5 m away. Correct responses in a session never exceeded 65% and averaged 46% on 617 trials. Without the blindfold, the seal's performance was almost 100%. The gray seal makes sound underwater but does not appear to use the clicks for echolocation in any way similar to the demonstrated capabilities of dolphins.

THURSDAY AFTERNOON, 10 NOVEMBER 1983

SENATE/COMMITTEE ROOMS, 1:00 P.M.

Session MM. Underwater Acoustics V: Signal Processing I (Précis-Poster Session)

Arthur B. Baggeroer, Chairman
Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Chairman's Introduction—1:00

Invited Papers

1:05

MM1. Broadband signal selection and generation for channel sounding. T. G. Birdsall (University of Michigan, 262 Cooley Bldg., Ann Arbor, MI 48109)

Two classes of signals and applications are considered. The T class, for deep water work where ray paths are an appropriate model, requires a processed output waveform that is a short pulse with zero amplitude beyond a

small neighborhood of the peak. The F class, for shallow water work where mode theory is an appropriate model, or for multioctave work where propagation varies significantly across frequency, requires a definite power spectrum. Although related, the two types of requirement are different in practice. Appropriate signal processing that balances signal-to-noise ratio and resolution will be discussed; it is called "factor-invert-matched-filtering." A specific measure of spectral flatness, together with in-band fractional power, are combined to yield a signal efficiency measure. Emphasis is placed on continuous periodic signals. The T-class signals to be discussed are binary, digital, phase-modulated by m sequences; variations are augmented m sequences and frequency modulation. The F-class signals to be discussed are linear FM and a particular digital discrete FM signal called PDFM.

1:30

MM2. The importance of correlators. G. Clifford Carter (U. S. Naval Underwater Systems Center, New London, CT 06320)

A tutorial review of generalized cross correlators and their importance to passive sonar signal processing will be presented. A discussion of the effects bandwidth, and integration time as well as multipath and multitarget interferences will be included. The presentation will emphasize broadband aspect of correlation processing including degradations experienced by narrow-band components. Fundamental limits on performance will be discussed.

Contributed Papers

1:55

MM3. Broadband signal processing in communication systems. Stanley L. Adams (Harris Corporation, P. O. Box 37, Melbourne, FL 32901) and Michael H. Brill (Science Applications, Inc., 1710 Goodridge Drive, P. O. Box 1303, McLean, VA 22102)

The broadband characteristics of the underwater acoustic propagation channel determine both the approach to communicating and the performance which can be achieved. This paper presents a detailed broadband characterization of the channel and uses this characterization to describe approaches to communication. Performance bounds are calculated and compared to channel capacity results.

1:59

MM4. Wideband, frequency-domain beamforming for coherent signal processing. Mark E. Weber (Code 5160, Naval Research Laboratory, Washington, DC 20375) and Rodney Heisler (Walla Walla College, College Place, WA 99324)

Coherent detection processors for sonar (e.g., matched filters for active systems) require continuous time series as input and have conventionally been associated with delay-and-sum time-domain beamformers. As an alternative, we have extended frequency-domain beamforming techniques to retain the coherence of wideband signals and have demonstrated efficient software for computing continuous time-series beams for flexible postprocessing. Using computer-generated "acoustic-array" data, the beamformer was shown to preserve the envelope, spectrum, and correlation properties of signals while yielding near theoretical performance in reducing beam side-lobe levels. Advantages over a conventional delay-and-sum beamformer are: (1) elimination of the need for high input sampling rates to achieve acceptable beam patterns—the frequency-domain approach is insensitive to the sampling rate provided it exceeds the (bandpass) Nyquist rate; (2) reduction of high-beam side-lobe levels caused by malfunctioning array elements—the response of missing array elements is readily estimated in the frequency domain by interpolating from the spectra of neighboring sensors; (3) minimization of hardware requirements—all stages of signal processing (bandpass filtering, holefixing, beamforming, time-line reconstruction, and matched filtering) can be efficiently performed in a general purpose array processor.

2:03

MM5. Frequency spreads of broadband signals reflecting from a random surface. Michael H. Brill (Science Applications, Inc., 1710 Goodridge Drive, P. O. Box 1303, McLean, VA 22102), Xavier Zabal (TRW, 7600 Colshire Drive, McLean, VA 22102), and Stanley L. Adams (Harris Corporation, Box 37, Melbourne, FL 32901)

Extending previous time-spread work by the present authors [J. Acoust. Soc. Am. Suppl. 1 73, S11 (1983)], the frequency spreading of an acoustic signal reflecting from an ocean bottom (in the presence of source and receiver motion) is computed using a simple geometric-acoustic model. In that model, a ray reflects specularly from each bottom facet, and arrives at the receiver if and only if the position and slope of the facet provide the requisite specular path. The model provides a connection between the expected power flux into the receiver aperture from a particular facet and the slope distribution of the bottom (assumed constant in depth). For a cw signal, the frequency spread can be computed by histogramming in $Q \equiv (f - f_0)/f_0$ the expected relative energy from a grid of bottom facets. (Here, f_0 is the frequency emitted at the source, and f is the frequency of the received signal.) The quantity Q depends only on the position of the facet, range, depth, and source and receiver velocities: it does not depend on f_0 , so long as the ray paths are independent of frequency, as in the present model. In fact, the expression for Q is the time derivative of delay time for a broadband signal, as well as having a narrow-band interpretation. Computed frequency-spread curves are presented for several source/receiver geometries. Because relative energy vs Q is a function of the channel and not of the sent signal, it may help to connect the broadband scattering function with the expected distribution of power in time and frequency from an input power impulse.

2:07

MM6. Spread signal processing. Andrew C. Callahan (Raytheon Company, Submarine Signal Division, Portsmouth, RI 02871)

The signal processing of broadband data often results in signals which are still spread over a region of parameters, such as, frequency, Doppler, delay, etc. It is of great interest to determine the possible gains achievable by the inclusion of all signal contributions into the formation of a region statistic. This is formulated into the analysis of distribution mixtures which is referred to as spread signal processing. To determine the potential gains in this approach, a particular example of signal plus noise and noise distributions were analyzed for detection performance as a function of signal level and the fraction of signal samples. Simulation results indicate an improvement over both the greatest sample and constant false alarm rate statistics for intermediate values of the signal sample fraction.

2:11

MM7. A study of the statistical properties of broadband signals. Fred W. Machell and Clark S. Penrod (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

A common assumption in the processing of broadband acoustic signals is that the underlying processes are Gaussian. Although Gaussian

processes arise naturally in the ocean acoustic environment, another motivation for the Gaussian assumption is the desire for analytically tractable results and simple algorithms. In this paper, a broadband propagation model is used to analyze the statistical properties of random signals propagated in a deep ocean environment. In particular, measurements of the first-order probability density functions associated with the source and received signals will be presented for several source-receiver configurations. These measurements provide insight into the appropriateness of a Gaussian assumption in broadband processing. They also illustrate the effects of acoustic propagation mechanisms on the statistical structure of a signal. Measurements of correlation functions are presented to show the effects of multipath propagation. [Work supported by the Office of Naval Research.]

2:15

MM8. A precision method of determining wide-bandwidth pulse delay in highly dispersive media. Herbert R. Carleton and Howard Austerlitz (College of Engineering and Applied Science, SUNY at Stony Brook, Stony Brook, NY 11794)

The high attenuation of certain classes of materials result in extensive distortion of the transmitted pulse due to dispersion. A system has been developed which enables the measurement of the attenuation and phase of highly absorptive materials over the frequency range of 0.1 to 20 MHz. Using a high-speed data acquisition system and a dedicated computer, the delay and dispersion component of the FFT transmission spectrum are separately computed by using a series of Hilbert transforms of the spectrum. The minimum delay component is first computed and subtracted from the phase spectrum. The resulting phase data are then used to determine the pure delay component to a high degree of accuracy. Experimental results will be presented on both model systems and on materials with attenuations as high as 100 dB/cm and dispersion as high as 20%.

2:19

MM9. Determination of the broadband spectra for saturated parametric sonars. R. J. Wyber^a (Applied Research Laboratories, The University of Texas at Austin, P. O. Box 8029, Austin, TX 78712-8029)

The determination of the propagation of a finite amplitude wave by Burger's equation is well known. Application of this solution to parametric sonars generally involves modeling a plane wave in the nearfield and a spherical wave in the farfield. The transition between these regions is not well defined and generally involves an arbitrary choice of a transition region. To avoid this requirement and to allow the solution for a broadband parametric sonar to be extended to the finite amplitude case, a model has been developed based on the approximation that the nonlinear losses in the fundamental are due entirely to the work done on the nonlinear source function. The model developed is similar in principle to that of H. Merklinger [J. Acoust. Soc. Am. 54, 1760-1761 (1973)], except that the *ad hoc* estimate of the nonlinear source function is replaced by an integral of the primary intensity. This solution is extended to allow for broadband source signals up to 1-oct bandwidth. A solution is obtained for the spectrum of the total power in the primary beam, which is independent of the cross-sectional area of the beam and is thus valid in both the nearfield and farfield.^a The author is on attachment from R. A. N. Research Laboratories, P. O. Box 706, Darlinghurst, Australia 2010.

2:23

MM10. Spectral high-pass filtering effects of broadband spatial normalization. Robert Perl (Bolt Beranek and Newman, Inc., 1300 N. 17th Street, Arlington, VA 22209)

Broadband energy detection is dependent upon a reliable estimate of the noise background. The energy at the output of a beam of an acoustic array is usually compared to the mean energy of its neighboring beams, and a detection is declared if a large discrepancy exists. This process of spatial normalization is equivalent to a high-pass filter which attenuates those spectral components which are substantially smaller than the design frequency of the array. This effect arises because of the substantial beam overlap at lower frequencies which precludes a totally independent esti-

mate of the background. The spatial normalization high-pass filtering effect has been calculated for a variety of normalization schemes for linear and circular arrays. In general, the attenuation of the filter is 6 dB/oct at frequencies much smaller than the design frequency of the array. Specific frequency response curves are presented, and the ramifications to broadband energy detection are discussed. These include modifications to the design of the Eckart filter, and the reduced noise bias effects from anisotropic remote shipping.

2:27

MM11. Techniques for isolating resonances in experimentally obtained broadband acoustic echoes. S. K. Numrich, N. H. Dale, and L. R. Dragonette (Code 5132, Naval Research Laboratory, Washington, DC 20375)

Theoretical descriptions of the scattering behavior of solid and hollow elastic bodies submerged in water have been given in terms of the form function of the body. These computations, usually carried out for a wide range of ka or frequencies, constitute a broadband analysis of the target's response. When performed in spherical or cylindrical geometries, the results of the computation can be separated into the rigid and purely elastic (or resonance) terms. Techniques have been developed for performing similar separations in experimentally obtained echoes for elastic targets. Of particular interest are the responses from bounded finite bodies for which no analytical model exists. Illustrations will include backscattered returns from solid and hollow targets not amenable to analytical solution. In the case of the hollow targets, plate waves generated by the incident sound will be identified using these broadband methods of target interrogation and analysis.

2:31

MM12. Performance of a broadband cross correlation detector. Peter S. Tong (Planning Systems Incorporated, 7900 Westpark Drive, Suite 600, McLean, VA 22102)

Cross correlating the outputs of two spatially separated sensors is one way to implement a broadband passive detector and bearing estimator. In the absence of targets, the correlation function will be nearly zero for all delays, and in the presence of a target, this function will peak at the delay corresponding to the target bearing provided the target is motionless. If the target is moving, the signal received at the two sensors will experience different amounts of time compression, and as a result the correlation function will not reach its peak value. Exact expressions for this degradation are obtained in terms of target geometry, target velocity, signal frequency, bandwidth, and coherent integration time. These expressions, although adequate for performance computation in a given situation, are too complex for system trade-off studies. Greatly simplified expressions are derived in this paper which give valuable insight into the relationships of the many variables and are more suitable for system design and evaluation.

2:35

MM13. A processing system for acoustic reciprocal transmission experiments. H. Nguyen and H. A. DeFerrari (Division of Ocean Engineering, Rosenstiel School of Marine and Atmospheric Science, University of Miami, Miami, FL 33149-1098)

Broadband signals such as linear maximal pseudorandom sequences (LMPRS) find many applications in acoustic remote sensing in the ocean. We have designed a system which both transmits and receives LMPRS entirely under microprocessor control. Timing of the experiments, generations of the transmitted signals, and coherence sampling of the received signal are accomplished with a low-power-consuming RCA 1802 processor. Parameters of the experiment can be set by software making possible a variety of experiments without hardware modifications. The system is battery powered and is designed to operate for periods of months at sea. Three systems have been constructed for acoustic reciprocal transmission experiments in the Florida Straits and are now being tested. The system will be described along with preliminary results from experiments. [Work supported by ONR and NOAA.]

MM14. Bandlimited cross correlation in a multipath environment: Simulation study results. Peter D. Herstein, Helen M. Sternberg, Bernard F. Cole, and Henry Weinberg (Naval Underwater Systems Center, New London, CT 06320)

The technique of intersensor cross correlation of acoustic signals is used to obtain such information as the spatial coherence of the environment or the position parameters of a broadband source. In general, acoustic energy propagates from a source to a sensor along multiple raypaths. However, cross-correlation performance is frequently analyzed in terms of a single wavefront (planar) model. For the planar model, the signal received at each sensor is approximated as the sum of the delayed and attenuated source signal plus uncorrelated Gaussian noise. This enables the use of performance criterion based on receiver signal-to-noise ratios. Extensive analysis of cross-correlation performance under the assumption of the planar model has been reported [G. C. Carter, *IEEE Trans. Acoust. Speech Signal Process.* ASSP-29(3) (1981)]. In this paper, modeling of the cross correlation between two sensors is extended to analyze the situation of a bandlimited signal propagating in a multipath environment. Using the generic sonar model (H. Weinberg, NUSC TD 5971C), a representative deep ocean configuration was simulated in which the propagation was dominated by four first-order bottom bounce type rays. Results are presented as functions of (a) the range and bearing of the source, and (b) the vertical tilt of the baseline of the sensors. It is shown that multipath effects can profoundly effect the level of the normalized cross-correlation peak. Signal-to-noise ratio at each sensor is found to be an inadequate estimator of correlation performance for this type of multipath environment. [This work was sponsored by the Naval Sea Systems Command, Washington, DC.]

MM15. Energy partitioning and optical-to-acoustic conversion efficiency during laser generation of underwater sound. Allan D. Pierce (School of ME, Georgia Institute of Technology, Atlanta, GA 30332)

Two mechanisms (thermoacoustic and ablative) are considered with regard to how much optical energy is eventually transformed into sound. The analysis of the former begins with a fluid dynamic analysis of the generation of entropy and acoustic modal fields taking into account viscosity and thermal conduction. The acoustic modal field satisfies the Westervelt-Larson inhomogeneous wave equation while the entropy modal field satisfies a transient heat conduction equation with similar source term. The derivation demonstrates that the Westervelt-Larson equation is valid under much broader circumstances than was previously asserted by Bozhkov and Bunkin. The bulk of the absorbed optical energy goes into the entropy mode, but the acoustic portion can be enhanced if the spatial and temporal dependence of the energy deposition resembles a solution of the wave equation, such that initially generated waves are continually pumped by the source. An upper limit to the conversion efficiency is of order $(\beta/\rho c_p)p_{max}$, where β is coefficient of thermal expansion and p_{max} is maximum attained (by source pumping) acoustic pressure in the source region. Ablation generates sound because of the recoil force acting on the surface when evaporated material jets off; the corresponding acoustic radiation efficiency is estimated using a model proposed by Bunkin *et al.* [*JETP Lett.* 13, 341-343 (1971)]. [Work supported by ONR.]

MM16. Pattern recognition by means of a hybrid adaptive autoregressive processing system. Bonnie Schnitta-Israel (KLD Assoc., Inc., East Hampton Field Office, 94 Pantigo Road, E. Hampton, NY 11937)

An automatic pattern recognition algorithm, consisting of event detection, feature extraction, and a decision process, was developed. The complete processing system was labeled SIGNET. The feature extraction aspect of SIGNET formulated the autoregressive (AR) coefficients from a hybrid adaptive AR algorithm in combination with a weighted linear threshold element into a linear prediction residual (LPR). The structure of the data sequence was then identified by extracting those LPR segments, which established event type boundary phenomena. SIGNET was evalu-

ated on two sets of data. The first set was comprised of nine independent underwater transient sources. The percent of correct recognition of SIGNET in that evaluation ranged from 93 to 98. The second data set was from a strong impulsive seismic source. The percent of correct recognition was 97. The mathematical foundation of SIGNET was also used successfully as a basis for a multiple source identification technique. [Work supported by ONR/Environmental Sciences.]

MM17. Passive source ranging by using mode filtering and mode phase comparing in waveguide. E. C. Shang, C. S. Clay, and Y. Y. Wang (Department of Geology and Geophysics, University of Wisconsin-Madison, Madison, WI 53706)

The conventional beamforming technique could not be used for source locating in waveguide owing to the modal interference structure of the field. In this paper a new method of passive source ranging in a layered waveguide has been proposed. The mode-filtering processor was used to process the field data of a vertical array in order to obtain individual modes. The source range information can be extracted by measurement of three individual mode phases. The source range was expressed in terms of the "mode interference distance" as following: $r = L_y D_y + \Delta r_y$; $\Delta r_y = D_y (\delta\phi_y/2\pi)$, where L_y is a certain integer; $\delta\phi_y$ is the phase difference of the i th mode and the j th mode; D_y is the "mode interference distance" defined by: $D_y = 2\pi/(k_i - k_j)$; k_i is the wavenumber of the i th mode given by a numerical mode code. The information of L_y can be estimated by means of comparing the phase of the j th mode with another mode, say m th mode, and then L_y was estimated by solving the following equation: $(\delta\phi_y/2\pi) = \text{Fractional part} \{ (D_y/D_{ym}) [L_y + (\delta\phi_{ym}/2\pi)] \}$.

MM18. Models of fluid structure interactions to optimize acoustics performance of sonar systems. Jacqueline Larcher, Jean-Marc Parot, and Jean-Paul Berhault (Société METRAVIB, BP 182, 69132 Ecully Cedex 2, France)

In sonar systems, the interaction of transducer elements with the inner dome cavity and the boundary layer with the dome structure affect the overall acoustic performance of the system in terms of aberrations and self-noise. Simple mathematical models have been derived to represent both phenomena. For acoustic aberrations resulting from coupling between the transducer and the acoustic cavity plus dome structure, an analytic numerical model has been used to identify significant technological parameters controlling the performance. A statistical energetic model describes coupling between the hydrodynamic boundary layer, the dome dynamic structural behavior and the inner sonar cavity. Experimental data have been collected on full scale and model sonar structures representing realistic excitation and loads conditions. Results of theoretical models and experimental data show good agreement on the effects of cavity modes and area mass for acoustic aberrations and of dome structural dynamic properties for hydrodynamic self-noise.

MM19. A comparison of broadband sonar discrimination between human listeners and a filter bank model. Douglas W. Martin and Whitlow W. L. Au (Naval Ocean Systems Center, P. O. Box 997, Kailua, HI 96734)

Broadband sonar discrimination capabilities of human listeners as a function of signal-to-noise ratio were compared with the constant-Q filter bank model of Chestnut, Landsman, and Floyd [*J. Acoust. Soc. Am.* 66, 140-147 (1979)]. A 48-s pulse (122-kHz peak frequency, 39-kHz 3-dB bandwidth) was used to collect echoes from aluminum, glass, and bronze cylinders. Discrimination experiments were performed in pairs, aluminum-glass, hollow-solid aluminum, and aluminum-bronze. A modified method of constants with the noise levels randomized in ten-trial blocks and the signals time-expanded into audio frequencies were used in the human experiments. Thirty contiguous filters covering a frequency band of 50 to 200 kHz were used in the constant-Q model. The human listeners performed significantly better than the computer model, requiring at least 5 dB less signal-to-noise ratio at the 75% correct threshold. Listeners reported that dominant cues occurred in the time domain.

MM20. Dependence of array gain on aperture angle for cylindrical and spherical arrays of directional sensors. Henry Cox and J. Michael Steele (Bolt Beranek and Newman Inc., 1300 North 17th Street, Arlington, VA 22209)

When receiving arrays are placed on air-backed structures in water, the effect of the structure or baffle is to alter the response of the hydrophones from their free field characteristics. The exact response character-

istics are complicated and frequency dependent. Typically the response is reduced near grazing incidence. When a beam is formed for maximum response in a selected direction, increasing the aperture to include elements for which the selected beam direction is near grazing is inefficient in increasing array signal gain and may actually lead to a loss in signal-to-noise ratio. The array gain dependence upon aperture angle for cylindrical and spherical arrays is presented for parametric families of effective element directivities. Both uniform and optimum shadings are considered. Results are presented in the form of parametric curves which can be used in system design tradeoffs.

THURSDAY AFTERNOON, 10 NOVEMBER 1983

SUNSET ROOM, 1:30 P.M.

Joint Meeting of 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 Standards Committee S1 on Acoustics

T. F. W. Embleton, Chairman S1

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

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

Meeting of Standards Committee S3 on Bioacoustics

W. A. Yost, Chairman S3

National Science Foundation, Sensory Physiology & Perception, Washington, DC 20550

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

THURSDAY AFTERNOON, 10 NOVEMBER 1983

DEL MAR/HELIX/SANTA FE ROOMS, 2:00 to 5:20 P.M.

Session NN. Musical Acoustics VI: Music Perception and Cognition

Diana M. Deutsch, Chairman

Center for Human Information Processing, University of California, San Diego, La Jolla, California 92093

Chairman's Introduction—2:00

Invited Papers

2:05

NN1. Dichotic listening to musical sequences: Relationship to hemispheric specialization of function. Diana Deutsch (Center for Human Information Processing, University of California, San Diego, La Jolla, CA 92093)

It is generally assumed that patterns of ear advantage in dichotic listening reflect greater involvement of the hemisphere contralateral to the preferred ear. This assumption is critically reviewed with respect to musical materials. A set of experiments is described which demonstrate the presence of a left-right anisotropy in the perception of frequency combinations. When two frequencies are simultaneously presented in a sequential setting, perception of these frequencies is superior when the higher is to the right and the lower to the left than when the higher is to the left and the lower to the right. Localization is also more accurate when the higher frequency is to the right and the lower to the left. These phenomena are shown to give rise to apparent patterns of ear advantage, whose direction may vary depending on such factors as overall error rate and listening strategy. Such apparent ear advantages should not be taken as reflecting greater involvement of the contralateral hemisphere. Implications for patterns of ear advantage reported for speech materials are also discussed. [Work supported by NIMH.]

2:35

NN2. Neuropsychology of music: A critical review of past work and future perspectives. Oscar S. M. Marin (Department of Neurology, Good Samaritan Hospital and Medical Center, Portland, OR 97201)

Localized brain lesions may result in disorders of music perception, recognition, reading, or may alter vocal or instrumental performance. Amusias appear in this way to be disorders in the field of music equivalent to those encountered in aphasias, agnosias, or apraxias in the areas of language and speech, visual perception, and general motor skills. Thus, amusias pose the same problems with respect to hemispheric specialization, and localization of function. The relationship between nervous structure and function, brain localization or specialized brain processors to a particular behavior depends largely on the cognitive structure and characteristics of the system on which such behavior is based. Thus it depends, for instance, on how constant, stable, or universal the cognitive system is; it depends on the sensory modalities on which it is based, on its requirements for sensory perceptual analysis, on its higher cognitive codes or categories, on its use of information storage systems, syntactic algorithms, and linkages to other systems. In light of these factors, a review of the clinical literature of the various types of amusias will be presented. Parallelisms or contrasts with equivalent disorders of speech and language, and visual perceptual disorders will be discussed.

3:05

NN3. Relative and absolute pitch perception by birds. Stewart H. Hulse (Department of Psychology, The Johns Hopkins University, Baltimore, MD 21218)

European starlings (*Sturnus vulgaris*) were trained to discriminate a class of rule-based, four tone ascending pitch patterns from a comparable class of descending pitch patterns. Then a series of transfer tests examined the birds' ability to maintain the discrimination under various transformations of the original pitch stimuli. The birds performed well when shifts in tone height occurred to novel exemplars within the original pitch training range, but not when shifts occurred to novel exemplars outside that range. When information about the direction of pitch change was reduced by shortening the patterns, the birds could solve the discrimination on the basis of the first two tones in a pattern, although performance improved as pattern length and, therefore, amount of information increased. The same series of transfers showed that in producing accurate discrimination the birds were using pitch cues based on *both* an absolute and relative perception of pitch. The data have implications for a comparative study of information processing in pitch perception.

3:35

NN4. Music of the animals. Edward C. Carterette and Carl Shipley (Departments of Psychology and Physiology, and The Brain Research Institute, University of California, Los Angeles, 405 Hilgard Avenue, Los Angeles, CA 90024)

We examine the question and some possible answers of whether animals make and appreciate music. We take as the material of music such aspects as pitch, intensity, timbre, texture, harmony, frequency modulation, and duration as well as rhythm which is just the interaction of all these aspects in structured patterns. Our answer is that a lot of God's chillun got rhythm.

4:05

NN5. Perception of rhythmic structures. I. J. Hirsh (Washington University and Central Institute for the Deaf, St. Louis, MO 63110)

The concern of auditory research with pitch and spectrum has led to an emphasis on individual phonemes in speech and on notes in music. Even the sequential properties of transitions in speech or of melody in music have emphasized changes in spectrum. While rhythmic structures often are aspects of such patterns, they may be studied in a "pure" form. The rhythmic grouping reported since the late 19th century was an interesting observation, but did not lead much further. Also, early studies on listeners' hearing of rhythmic patterns may have been much affected by an imitative, tapping response, accompanied by characteristics of a motor system. Seashore, in his test of musical aptitudes, studied rhythmic perception through a same-different procedure but did not exploit the technique toward notions of stimulus complexity, or good or easy structures. Whether such pattern properties will accord with Martin's syllabic structures, or Garner's groups of runs and gaps, remains to be worked out by studies on recognition and discrimination. Data from a study along these lines will be discussed. [Work supported by NINCDS.]

NN6. **The role of auralization in pitch or tonality recognition.** W. Dixon Ward (Hearing Research Laboratory, University of Minnesota, 2630 University Avenue SE, Minneapolis, MN 55414)

Previous research [E. Terhardt and W. D. Ward, *J. Acoust. Soc. Am.* 72, 26-33 (1982)] has shown that most trained pianists can tell when a particular selection from Bach's "Well-Tempered Clavichord" has been shifted upward or downward in key from the original signature, performing at a level significantly better than chance even when this transposition is only one semit. This result implies that these persons possess some form of absolute pitch, or absolute tonality, even though they claim little or no ability to name isolated pitches. The degree to which this recognition process depends on auralization (anticipation of how the excerpt should sound) arising either from the oral announcement of the correct key that precedes each excerpt or from the (untransposed) visible score on the answer sheet was examined by removing these cues. Performance declined slightly but remained above chance, indicating that auralization, while helpful, is not essential. So unless there is a "best" key for any given piano composition, and Bach unerringly chose it, these pianists' recognition of transposition was based at least partly on long-term memory of the excerpts as heard in the original key. [Research supported by the Bryng Bryngelson Communication Disorders Research Fund.]

Contributed Paper

5:05

NN7. **Frequency generalization constrained by pitch processing in the European starling?** Jeffrey Cynx (Department of Psychology, The Johns Hopkins University, Baltimore, MD 21218)

Earlier work with pitch processing has shown that the European starling (*Sturnus vulgaris*)—unlike humans—fails to transfer learned pitch discriminations out of the original frequency range (S. Hulse, J. Cynx, and J. Humpal, "Effects of shift in pitch context on serial pitch perception in birds," in preparation). The present work was undertaken to determine if

the failure to generalize is due to the simultaneous active processing of pitch or the inability to make generalizations from simple frequencies. Four wild-caught adult starlings were trained to peck at a lit key only on trials during which a 2000-Hz tone was absent. Once discrimination was achieved, probes were introduced every other session at $p = 0.50$ to determine generalization from 1000 to 4000 Hz. Findings are in agreement with earlier work, using a Pavlovian procedure, which indicated a range constraint for frequencies [J. E. Trainer, "The auditory acuity of certain birds," unpublished doctoral dissertation, Cornell Univ. (1946)], but also show that the constraint is interactive with the processing of relative and absolute pitch information.

THURSDAY AFTERNOON, 10 NOVEMBER 1983

CHAMBER ROOM, 2:00 TO 5:00 P.M.

Session OO. Engineering Acoustics V: Transients

Peter R. Stepanishen, Chairman

Department of Ocean Engineering, University of Rhode Island, Kingston, Rhode Island 02881.

Chairman's Introduction—2:00

Invited Papers

2:05

OO1. **The physics of acoustic transients.** Miguel C. Junger (Cambridge Acoustical Associates, Inc., 54 Rindge Avenue Extension, Cambridge, MA 02140)

Many if not most sound fields, whether natural or man-generated, desirable or objectionable, are too transitory to be approximated with steady-state solutions. Furthermore, transient pulses are used increasingly to explore the frequency dependent characteristics of waveguides, geological formations, and echo-ranging targets. This shifts the burden from effectively steady-state measurements at various frequencies to signal processing of transient signals, an evolution facilitated by today's computer capability. In spite of their importance, transients are effectively ignored in standard text books with the exception of application-oriented fields such as room acoustics and loudspeaker theory. This review paper deals with basic phenomena associated with impulsively accelerated and decelerated boundaries. Transient radiation loading and farfields are described for planar and convex pulse generators. Energy partition between incompressible nearfield and acoustic farfield is illustrated. Finally, requirements are formulated for the equivalence of transient and cw scattering measurements.

2:30

OO2. **Time domain methods of investigating acoustic transient phenomena in arrays.** P. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881)

A review of linear acoustic transient phenomena in active arrays is presented. Such transients occur as a result of the finite bandwidth of the elements, the time delay caused by the propagation effects in the medium, and acoustic interaction effects among the elements. The characteristics of acoustic radiation from a time dependent monopole and dipole sources are first discussed to address such phenomena. These results are then extended to distributed sources where the importance of edge effects is clearly observed and discussed. Retarded potential methods to account for the effects of boundaries on the transient acoustic field are then presented. Such methods led to the development of impulse response techniques of evaluating acoustic transient phenomena. The impulse response technique is discussed in detail and numerical results are presented to illustrate transient acoustic pressure field and element interaction phenomena in active arrays.

2:55

OO3. Transient effects in acoustic scattering and radiation problems. G. C. Gaunard (Naval Surface Weapons Center, R43, Silver Spring, MD 20910)

We consider examples of acoustic radiation and scattering problems that we have worked out over the years where transient effects are investigated. In chronological order, we first study the sound radiation emitted by (multipole) point sources of various types, undergoing accelerated motions along various rectilinear, circular, or helicoidal paths. Changes in source acceleration results in an emission of a transient burst of radiation, and in a "headlight" effect, that we have quantitatively examined by techniques similar to those developed by J. Schwinger for accelerated electrons. We then analyze the scattering of a sound δ -pulse incident on a rigid sphere. When the transient pulse hits the sphere, it gives it its momentum and makes it undergo an impulsive acceleration in the same direction by radiation pressure. The so-called Kirchhoff solution [G. R. Kirchhoff, *Mechanik* (Teubner, Leipzig, 1883—one hundred years ago!) 3rd. ed., pp. 316–320] for the radiation field from an impulsively accelerated sphere is shown to exactly coincide with the first-order terms of the complete scattered field expression that we have obtained via the Resonance Scattering Theory (RST). This exhibits very enlightening connections between transient solutions of radiation and scattering problems.

3:20

OO4. Transient acoustic sources. John V. Bouyocous, David E. Nelson, and Roger L. Selsam (Hydroacoustics Inc., P. O. Box 23447, Rochester, NY 14692)

The characteristics of several different classes of transient, underwater acoustic sources are set forth and compared. Two relatively new sources—the water gun and imploding piston source—are compared with more well-established explosive devices and air guns. Characteristics examined include spectral features, efficiency of energy conversion, peak source levels, size, weight, and depth dependence. In addition, the performance advantages and limitations of the various source types in multi-source arrays is discussed.

Contributed Papers

3:45

OO5. Transient sound radiation from a clamped circular plate—Impulse response method. Adnan Akay and Michael Latcha (Mechanical Engineering Department, Wayne State University, Detroit, MI 48202)

A theoretical analysis of transient sound radiation from a clamped circular plate is obtained using a pressure impulse response method. The vibration response of the plate to a transient point force is obtained. The modal pressure impulse response functions for the plate are derived from the Rayleigh surface integral and numerically convoluted with the modal acceleration response of the plate. The impulse response functions are closely related to the mode shapes and the geometry of the problem. They relate the spatial domain to the temporal domain of the pressure waves. The pressure impulse response waveforms are given for a number of plate modes and the changes in the waveforms with distance from the plate are shown. Sound radiation due to forced and free vibrations of the plate are discussed. The results are compared with those obtained by direct numerical integration of the Rayleigh surface integral and the experiments. [Work supported by WSU and NSF.]

4:00

OO6. Interpolation techniques for sound radiation prediction from a flat vibrating surface. Pranab Saha,¹ and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

In an attempt to develop a model of flat surface vibrations to make accurate farfield sound radiation predictions, two interpolation techniques (12-point formula and 24-point formula) have been developed. The

purpose of these techniques is to generate additional surface vibration data at regions of sparse observations and in between gaps of existing data to obtain smooth surface vibration data. The development of the interpolation techniques is based on the equally spaced data points and the method of undetermined coefficients, given by the equation, $a_n(0,0) = a_n(x,y) - (k_1x + k_2y + k_3x^2 + k_4xy + k_5y^2)$, where $k_1, k_2, k_3, k_4,$ and k_5 are the coefficients needed for the best fit of the curve in order to relate the unknown surface vibration $a_n(0,0)$ with the known surface vibration $a_n(x,y)$. An expression for the relative error has also been derived. This expression provides information about the vibration distribution over the flat surface and hence about the applicability of the interpolation techniques. The results of the application of these two interpolation techniques based on an available set of test data are reported here, and are compared with the measured sound pressure levels in the farfield.

¹ Present address: C. F. Braun and Co., Chemical Engineering Department, 1000 South Fremont Avenue, Alhambra, CA 91802.

4:15

OO7. The Rayleigh distance and geometric nearfield size for nonplane sound radiators. Herbert L. Kuntz (Hoover Keith and Bruce Inc., 9730 Town Park, #108, Houston, TX 77036), Elmer L. Hixson (Department of Electrical Engineering, The University of Texas, Austin, TX 78712), and William W. Ryan, Jr. (Harding University, Searcy, AR 72143)

The geometric nearfield of a finite acoustic radiator is the area around the radiator where the sound pressure level does not follow spherical or cylindrical spreading. The geometric nearfield has been studied extensively for circular and rectangular piston radiators. The Rayleigh distance $D = ka^2/2$, where $k = 2\pi f/c$, f is the frequency, c is the sound

speed, and a is half the largest dimension, has been used to approximate the size of the geometric nearfield of plane radiators. We found that, for variously shaped radiators, the Rayleigh distance is useful in approximating the geometric nearfield size. As with plane radiators, we found large SPL changes with radial distance in the geometric nearfield. One implication is that in noisy environments, where the SPL around a large machine may, necessarily, be measured in the nearfield, the sound power contribution to the sound field can be grossly underestimated. Knowledge of the nearfield size may be useful in industrial situations for determining the validity of sound power estimates. [Work supported by the U.S. Coast Guard.]

4:30

OO8. Tow-powered low-frequency sound source. H. C. Schau, A. L. Van Buren (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856), and F. J. Radosta (Embry-Riddle Aeronautical University, Daytona Beach, FL 32014)

A novel method for production of low-frequency sound from a submerged rigid body with a clamped gas-filled hemispherical membrane is discussed. The device is driven hydrodynamically through the motion of a ship or other towing vehicle. Analytical models are developed for the device which includes both bending and membrane theories for the shell deformations with the inclusion of the pressures from the driving jet and

the acoustic radiation. Jet stability of the driving mechanism is discussed relative to the normal mode of vibration of the hemisphere. The hemispherical displacements are presented as a sum of resonant modes and forced modes which are in turn employed in a parametric study of optimizing acoustic output. Results include analysis of resonant vibration modes and prediction of acoustic radiation in both the nearfield and the farfield. Experimental data are compared with the model and future experiments are presented.

4:45

OO9. Diffraction at a fixed point on a fluid-loaded plate. R. V. Waterhouse and F. S. Archibald (David Taylor Naval Ship R&D Center, Code 194, Bethesda, MD 20084)

At the last meeting, some theoretical results were presented for the nearfield diffraction pattern formed by plane bending waves passing a fixed point on a thin elastic plate *in vacuo*. This case is a good approximation to a metal plate in air, but not to a metal plate with water on one or both sides. The latter case is of interest, for example, when the plate represents the hull of a ship, and in the present work the previous analysis is extended to this case. The fluid loading adds a third dimension to the problem, as now some of the acoustic energy is transferred from the plate to the water. The expression for the sound pressure contains a complicated integral which must be evaluated numerically. The integral contains a fifth degree polynomial in the denominator, the five roots of which contribute five poles to the integrand.

THURSDAY AFTERNOON, 10 NOVEMBER 1983

CALIFORNIA ROOM, 1:30 TO 4:35 P.M.

**Session D. Architectural Acoustics II: The Technical Committee on Architectural Acoustics
V. O. Knudsen Distinguished Lecture**

William J. Cavanaugh, Chairman
Cavanaugh Tocci Associates, 18 Main Street, Natick, Massachusetts 01760

Chairman's Introduction—1:30

Invited Paper

1:35

D1. Some reflections on room acoustics. Cyril M. Harris (Columbia University, New York, NY 10027)

This paper reviews selected aspects of the field of room acoustics, emphasizing the contributions of V. O. Knudsen and his close associates and former students. The work done by the author with Knudsen in establishing so-called "optimum" reverberation time curves for "Acoustical Designing in Architecture" is discussed. Finally, the author describes what he considers the most important unsolved problem in room acoustics and gives suggestions for an approach that may be useful in its solution.

Session PP. Architectural Acoustics VI: Room Acoustical Design, Assessment, Modeling, and Simulation

R. Lawrence Kirkegaard, Chairman
R. L. Kirkegaard and Associates, 125 Everest Road, Lombard, Illinois 60148

Chairman's Introduction—2:30

Contributed Papers

2:35

PP1. An auditorium design based upon subjectively optimal acoustics for The Washington Center, Olympia, Washington, USA. Yoichi Ando (Faculty of Engineering, Kobe University, Kobe, Japan) and Dennis Noson (Noson & Associates, 2605 East Lynn Street, Seattle, WA 98112)

Recently developed acoustical design criteria, derived from an exhaustive program of subjective testing at Kobe University and the University of Göttingen, have been applied to the acoustical design of The Wash-

ington Center, a multipurpose performing arts facility in Olympia, Washington, USA. In response to the programmatic emphasis on high-quality acoustics for orchestral music, the architectural configuration progressed from a wide, single-balcony scheme to one that is narrower with two balconies. The acoustical design effort was concentrated on the size and shape of wall reflectors, room asymmetry, and diffusing elements. The first reflection time delay, binaural dissimilarity, and total energy of the sound field were predicted with a computer model of the auditorium. Adjustments were made to the shape, checked by computer, and synthesized into the architectural design, arriving at optimum values for selected seat locations.

PP2. Comparison of acoustic experience of studio recordings and live concerts. Alexandra G. Sotiropoulou and David B. Fleming (University College, London, England and Technical University of Athens, Athens, Greece)

It has been common in recent years to use recorded music in the investigation of subjective qualities of concert hall acoustics. The object of this paper is to compare experimental results from recorded music evaluations and live concert evaluations. A list of 54 opposite labels describing acoustic qualities of concert halls were used at the poles of bipolar rating scales in the evaluation of studio recordings. The raw judgments were analyzed by factor analysis and five independent factors were produced, namely body, clarity, tonal quality, extent, and proximity. The validity of these results was then tested in the environmental complexity of live concert conditions. To this end 27 scales were evolved to represent the previously obtained factors and were used at three live concert evaluations. Four to five independent factors were produced depending on the concert situation. Four of these factors, namely body, clarity, tonal quality, and proximity had also previously emerged; two factors which emerged only in the live concerts were spaciousness and intimacy [A. G. Sotiropoulou *et al.*, Proc. Inst. Acoust., Edinburgh (1982)]. These results show that there are independent sets of acoustic qualities (factors) common to the studio recordings used in this study and to live concerts.

3:05

PP3. Cliff Temple Baptist Church: Acoustics for worship and recording. Russell C. Campbell (General Dynamics, Fort Worth, TX and Department of Physics, Texas Christian University, Fort Worth, TX 76129) and Richard J. Lysiak (Department of Physics, Texas Christian University, Fort Worth, TX 76129)

Cliff Temple Baptist Church, located in Dallas, Texas, prides itself on its excellent music facilities which are not only used for worship, but also for recording RCA's Dallas Symphony series of digital Red Seal albums. Reviewers such as David Hall [Stereo Rev. 46, 76 (1981)] have highly praised the acoustics of Cliff Temple, as well deserved according to this study. Four types of objective data consisting of noise measurements, echograms, reverberation plots, and characteristic spectra, are used to determine sound field characteristics present in the hall. Using wide dynamic range analysis techniques [R. Campbell and R. Lysiak, Bull. Am. Phys. Soc. 28, 200 (1983)] greatly enhanced the measurement of acoustical parameters, giving insight into the subjectively perceived superb acoustical qualities. The objective data is then applied to subjective categories which are used for comparison of Cliff Temple Baptist Church with other facilities. [R. Campbell, M. S. thesis, Texas Christian University (1983).]

3:20

PP4. Determining the necessary number of reflections for an accurate image source simulation of room acoustics. Lawrence E. Zagar and James K. Thompson (Mechanical Engineering Department, Louisiana State University, Baton Rouge, LA 70803)

Previous attempts at using image source computer models to determine sound pressures in rooms or large enclosures have had mixed results. One evident feature of these studies is that the maximum number of reflections simulated from each room surface is critical. An investigation was conducted using measured data for three rooms and computer simulations to evaluate various means of predicting the necessary number of reflections for accurate modeling. Schemes have been presented in the literature for predicting the necessary number of reflections for specific levels of accuracy. These tests have indicated that such formulas are highly inaccurate. There have also been remainder formulas presented in the literature for correcting computations with a limited number of reflections. The remainder formulas currently available were found to be unable to accurately compensate for model errors in analyzing actual rooms. Where others had indicated a maximum of only three or four reflections need be considered, these investigators found that as many as 40 reflections may sometimes be required. More typically, ten to 20 reflections from each room surface must be modeled for acceptable results. The required number of reflections was found to be a complex function of room size, room proportions, and the distribution and amount of absorption.

PP5. Music performance acoustics and room shape: An investigation employing an images model of room acoustics. Nicholas A. Edwards (Artec Consultants, Inc., 254 Seventh Avenue, New York, NY 10001)

Critical listeners often observe that rectangular (shoe box) rooms have "better acoustics" for music than fan shaped rooms of similar size. In this paper we examine the relationship between room shape and some physical aspects of sound fields in the room. The tool for our analysis has been a geometric acoustics (images) model implemented on a microcomputer. Three-dimensional impulse responses have been computed for some simple fan-shaped, reverse-fan-shaped, and rectangular rooms. Various means of graphical display are used to point out those trends in the time and space distributions of the impulse response that are tied to the fundamental shape of the room. Lateralization of the sound field and development of the reverberant field are two acoustical processes in particular that have their roots in the architecture of the space. Drawing on the past two decades of research into the influence that the direction of arriving sound has on listener perceptions and preferences for music, we identify links between the "sound" of a room and its shape.

3:50

PP6. Mathematical and computational issues in the deconvolution of a finite-length wavelet from a room impulse response under noisy conditions. John P. Walsh (Artec Consultants Inc., 245 Seventh Avenue, New York, NY 10001, and Artec/Ars Nova Research, 230 West 15th Avenue, Vancouver, Canada V5Y 1X9)

The theoretical and computational issues which arise in the design and use of algorithms for digital deconvolution are discussed. Derivation of room impulse responses from measurements made using source signals which do not exhibit delta function behavior, under noisy conditions imposed by both the ambient conditions on-site and finite precision in the application of the deconvolution filter, form the focus of the paper. The genealogy of these methods in the literature of time series analysis, geophysical signal processing, and digital image restoration is described.

4:05

PP7. Acoustics of coupled spaces relative to assisted reverberation. Bruce E. Walker (Consulting and Research in Acoustics, 2659 Townsgate Road, Suite 101, Westlake Village, CA 91361) and Ludwig W. Sepmeyer (Consulting Engineer, 1862 Comstock Avenue, Los Angeles, CA 90025)

One method of providing variable reverberation in an auditorium is to couple it electroacoustically with an auxiliary reverberation room. Parameters available for control of the composite characteristics include microphone and loudspeaker locations in both spaces, the reverberation time in both spaces, and the gain of the coupling amplifiers. By solving the coupled differential equations representing a simplified model of the arrangement, it was found that a variety of decay "shapes" ranging from concave (rate decreasing with time) to convex could be realized. Reverberation time could be varied over a greater than two to one range, while maintaining a "desirable" straight characteristic. Results of computer simulations for steady-state and transient excitations will be displayed.

4:20

PP8. The decay of sound in rooms with amplification systems. Eugene T. Patronis, Jr. (School of Physics, Georgia Institute of Technology, Atlanta, GA 30332) and Ted Uzzle (Altec Lansing Corporation, Box 3113, Anaheim, CA 92803)

The regenerative behavior of sound reinforcement systems affects the sound level and interacts with the reverberative decay of sound in enclosures in which they operate, in ways not previously described. For a system of loop gain $A(s)B(s)$, the sound levels will be $-20 \log[1 - A(s)B(s)]$ (dB) above those predicted by classical feedforward methods. After the source of sound to be amplified is shut off, the regenerative loop will continue producing sound, and will take $P / -57.6 \log[A(s)B(s)]$ (s) to decay 60 dB, where P is the regenerative path length (m). Methods for calculating and measuring the decaying loop, by itself and in combination with the room's reverberation time, are given.

Session QQ. Psychological Acoustics V: Binaural Hearing and Lateralization

Judy R. Dubno, Chairman

Head and Neck Surgery, Medical Center, University of California, Los Angeles, California 90024

Chairman's Introduction—1:00

Contributed Papers

1:05

QQ1. Binaural interaction in hearing impaired listeners. Kaigham J. Gabriel (Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

The binaural hearing abilities of four sensori-neural loss listeners were tested in four experiments: (1) interaural time discrimination, (2) interaural intensity discrimination, (3) interaural correlation discrimination, and (4) binaural detection experiments. All tests were conducted on each subject using third-octave bands of noise centered at 250, 500, 1000, 2000, and 4000 Hz. "Binaural Audiograms" can be obtained by plotting the ratio of normal discrimination jnds to measured discrimination jnds versus frequency on a log-log scale. In general, the frequency dependence of binaural hearing loss, so defined, appears unrelated to the frequency dependence of hearing loss as measured by current monaural audiometric techniques. Furthermore, impairments of interaural time discrimination are independent of impairments in interaural intensity discrimination. Finally, the MLD alone was not related to loss of interaural time or intensity sensitivity. Two of the subjects tested showed elevated NoS π and NoSo thresholds, poor interaural time and intensity discrimination but a "normal" MLD. [Work supported by NIH Grant NS10916.]

1:25

QQ2. A general modification for models of binaural interaction. K. J. Gabriel, H. S. Colburn, P. M. Zurek, and N. I. Durlach (Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

Although many studies have demonstrated the relationship between lateralization and binaural detection, current models of binaural interaction are unable to predict performance in both tasks with the same set of values for the model parameters. By incorporating a mechanism which low-pass filters (averages) interaural time and intensity differences into a simple, narrow-band model of binaural interaction, results from interaural time and intensity discrimination tests can be used to predict correlation discrimination and NoS π detection results. Evidence for such a low-pass mechanism has most recently been reported by Grantham and Wightman [J. Acoust. Soc. Am. 72, 1178-1184 (1982)]. For both normal listeners and hearing-impaired listeners with a variety of different losses, the predictions of the augmented model suggest that binaural hearing in correlation discrimination and NoS π detection tasks can be characterized by a listener's sensitivity to interaural time and intensity differences. [Work supported by NIH Grant NS10916.]

1:45

QQ3. Lateralization of clicks based on interaural time: Additivity of information across frequency. Elizabeth M. Wenzel and Ervin R. Hafter (Department of Psychology, University of California, Berkeley, CA 94720)

Sensitivity to interaural differences of time was measured for trains of 1 to 32 (n) clicks presented at rates of 400 or 200 per second. The center frequencies of the bandpass clicks were either 4, 5.2, 6, or 7.2 kHz. As in earlier research, plots of log threshold versus log n produced functions

whose negative slopes grow more shallow with increasing rate. When sensitivity was measured for composite signals made of clicks whose center frequencies were 4 and 6 kHz, or 4, 5.2, 6, and 7.2 kHz, the slopes of the functions were similar to the single-frequency conditions. However, the intercepts for the two- and four-frequency stimuli were decreased by factors of approximately $\sqrt{2}$ and 2, respectively, relative to the case of single frequencies. These data demonstrate that the rate-dependent saturation of binaural information previously shown to operate within frequency channels [E. R. Hafter and R. H. Dye, J. Acoust. Soc. Am. 73, 644-651 (1983)] does not affect the integration of information across channels. [Supported by NIH.]

2:05

QQ4. The extension of binaural crosscorrelation modelling by a mechanism of lateral inhibition. Werner Lindemann (Lehrstuhl für allgemeine Elektrotechnik und Akustik, Ruhr-Universität, D-4630 Bochum, Postfach 102148, West Germany)

To model the stationary and dynamic behavior of the binaural system we present an extension of the well-known binaural crosscorrelation models which is based on a lateral inhibition mechanism along the crosscorrelation axis. The running interaural crosscorrelation is calculated by counting spike coincidences in frequency bands. The resulting neural activity distribution is fed into the lateral inhibition mechanism. The required EE and EI gates can be combined in different ways to optimize the performance of the model. A wide field of psychoacoustic lateralization and localization data can be explained very effectively. For example, interaural phase as well as interaural level differences are evaluated in order to model lateralization as well as time-intensity trading. For nonstationary signals the model can explain the law of the first wavefront and related psychoacoustic effects. The model parameters are matched to the psychoacoustic data by choosing optimal time constants of the spike firing process, the spike coincidence window, and the inhibition mechanism. Some model results will be presented graphically in the form of "neural activity patterns." [Work supported by the Deutsche Forschungsgemeinschaft.]

2:25

QQ5. Envelope-based lateralization of low-frequency waveforms. Leslie R. Bernstein (Department of Psychology, University of Illinois, Champaign, IL 61820) and Constantine Trahiotis (Department of Speech and Hearing Science and Department of Psychology, University of Illinois, Champaign, IL 61820)

Several recent investigations suggest that listeners either cannot or do not use interaural temporal disparities (ITDs) in the envelope of low-frequency waveforms in lateralization tasks [G. B. Henning, J. Acoust. Soc. Am. 68, 446-453 (1980); G. B. Henning and J. Ashton, Hearing Res. 4, 185-194 (1981); G. B. Henning, Hearing Res. 9, 153-172 (1983)]. Listeners in those studies may have been unable to process these ITDs principally because of the types of stimuli presented. We employed an acoustic pointing task in which listeners varied the interaural intensive difference of a 500-Hz narrow-band noise (the pointer) so that it matched the intracranial position of a second, experimenter-controlled stimulus (the target). Target stimuli were sinusoidally amplitude-modulated tones cen-

tered on 500 Hz or 1 kHz, and modulated at 25, 50, or 100 Hz. Targets were presented with either the entire waveform delayed or with only the envelope delayed. Our results suggest that delays of the envelope *do* affect the lateral position of our low-frequency targets. However, the envelope-based cues appear to interact with those provided by the fine-structure which appear to be dominant.

2:45

QQ6. A search for central lateral inhibition. W. M. Hartmann (Department of Physics, Michigan State University, East Lansing, MI 48824 and Institut de Recherche et Coordination Acoustique/Musique, 31 rue Saint Merri, 75004, Paris, France)

To understand the existence of the binaural edge pitch [Klein and Hartmann. *J. Acoust. Soc. Am.* 70, 51-61 (1981)] appears to require central lateral inhibition in the human auditory system. We have looked for this effect in central masking experiments. Using the binaural-edge noise, which creates the binaural edge pitch, as a masker, we found pulsation thresholds for sine tones in two frequency ranges where the binaural edge pitch exists. We further found masking level differences using the same pulsation threshold technique. Theoretically the difference between the binaural-edge thresholds and the MLD thresholds should show the peak and valley signature of lateral inhibition. No such structure was found. We suggest that this negative result does not exclude the possibility of central lateral inhibition, but that the time course of central lateral inhibition makes the pulsation threshold technique an inappropriate means for observing the effect. [Work partially supported by the NIH, grant NS 17917.]

THURSDAY AFTERNOON, 10 NOVEMBER 1983

FORUM ROOM, 1:30 TO 4:50 P.M.

Session RR. Physical Acoustics V: Rayleigh Waves, Leaky Waves, etc.

Bernhard R. Tittmann, Chairman

Science Center, Rockwell International, 1049 Camino Dos Rios, Thousand Oaks, California 91360

Chairman's Introduction—1:30

Contributed Papers

1:35

RR1. Quantitative analysis of pulsed pseudo-Rayleigh waves. J. H. M. T. van der Hijden (Schlumberger-Doll Research, P. O. Box 307, Ridgefield, CT 06877)

Acoustic pseudowaves (leaky waves) along a plane interface are studied in the space-time domain. Analysis of the Green's function as derived by the Cagniard-de Hoop method allows a quantitative description of the propagation of a pseudo-Rayleigh pulse along a fluid/solid interface. After applying the Cagniard-de Hoop method the answer has the form of a convolution of the input signal of the source with the explicitly obtained space-time Green's function ("system's response"), which clearly shows each feature of the time behavior of the acoustic quantities at different locations and their dependence on the material parameters involved. From an analytic study of the space-time Green's function for the reflected pressure a quantitative description of the pseudo-Rayleigh phenomenon has been derived. This includes a condition which limits the range of existence and an equation for the travel speed. A pseudo-Rayleigh factor which determines the relative strength of the phenomenon is defined and the decay of pulsed wave motion along the interface is studied. Using this factor it becomes clear why the pseudo-Rayleigh wave is very strong at a water/steel interface, while it does not exist on, e.g., a water/ice interface. Numerical results will be shown.

1:50

RR2. Rayleigh wave transmission reciprocity past a step change in elevation. Jacques R. Chamuel¹⁾ (The Charles Stark Draper Laboratory, Inc., 555 Technology Square, Cambridge, MA 02139)

Contradicting experimental, approximate theoretical, and numerical results have been reported in the literature by several authors [L. M. Martel *et al.*, *Bull. Seism. Soc. Am.* 67, 1277-1290 (1977)] on Rayleigh wave transmission coefficient past a step change in elevation. In this paper, reciprocity for both wave propagation direction across a step is demonstrated experimentally using a new ratio technique eliminating error sources introduced from elastic wave generation, coupling, and detection. Two-dimensional ultrasonic models were used to obtain the data. The experimental transmission coefficient measurements are compared with other published results. The ratio of step height to Rayleigh wavelength tested ranged from 0.159-1.33. The effect of step height on the interference of the transmitted wave components is examined. The conversion between Rayleigh and shear waves at the step plays an important role in the transmission coefficient. ¹⁾Present address: Sonoquest, P. O. Box 584, Sudbury, MA 01776.

2:05

RR3. New ultrasonic modeling findings on Rayleigh wave propagation and their implications. Jacques R. Chamuel¹⁾ (The Charles Stark Draper Laboratory, Inc., 555 Technology Square, Cambridge, MA 02139)

The presence of unexplained large attenuation of low-frequency components and inverted dispersion of Rayleigh waves crossing the ocean-continent margin, the Tibetan Plateau, and the Iranian Plateau are examples of complex unsolved problems on Rayleigh wave propagation across vertical boundaries. In this paper, new ultrasonic modeling findings on Rayleigh wave propagation across step changes in elevation using two- and three-dimensional models are presented explaining the previously

mentioned phenomena. Laboratory seismic ultrasonic modeling results on Tibet are presented matching actual field data. Physical insights into the cause of the inverted dispersion and attenuation are obtained from the reported results. Implications of the new findings indicate the need to reconsider the current ocean-continent margin, the Tibetan Plateau, and the Iranian Plateau crustal structure models. Opposite to the current belief of the presence of a low-velocity high-damping layer (possibly partially molten) near 30 km below the surface of the Tibetan Plateau [P. Bird *et al.*, *Nature* **266**, 162-163 (1977); W. P. Chen *et al.*, *J. Geophys. Res.* **86**, 5937-5962 (1981); K. Y. Chun *et al.*, *Bull. Seism. Soc. Am.* **67**, 735-750 (1977)] the new reported findings support the author's concept of a solid low attenuation solid crustal structure. A brief discussion on liquid/solid interface waves propagation past a step change in elevation will follow the Rayleigh wave results. ^{a)} Present address: Sonoquest, P. O. Box 584, Sudbury, MA 01776.

2:20

RR4. Reflection of narrow ultrasonic beams at fluid/solid interfaces. Finn B. Jensen and Henrik Schmidt (SACLANT ASW Research Centre, 19026 La Spezia, Italy)

The reflection of ultrasonic beams near the Rayleigh angle for a fluid/solid interface is a well researched subject. Existing theories, however, are all approximate with limitations such as beamwidth large compared to the wavelength, lossless media, parallel beams of particular shape (Gaussian), etc. A new numerical model has been developed to solve the reflection problem without any of the above limitations. This technique is based on the fast-field solution to the wave equation, and solutions can now be obtained for any beamwidth, and for parallel, diverging, and converging beams. Furthermore, due to the numerical efficiency of the computer code, pulsed-beam solutions can be generated in reasonable time using Fourier synthesis of the time signals. Computational results are presented for various beam configurations reflected at water/steel and water/aluminum oxide interfaces, and the results are compared with available experimental data.

2:35

RR5. Observations of backward leaky Lamb waves in plates. Michel de Billy,^{a)} Laszlo Adler (Department of Welding Engineering, Ohio State University, Columbus, OH 43210), and Gérard Quentin (Groupe de Physique des Solides, Université Paris, Paris 7, France)

We have previously reported [*J. Acoust. Soc. Am.* **72**, 1018-1020 (1982)] that a finite ultrasonic beam produces backward leaky-Rayleigh waves from liquid-solid interface. These backward propagating leaky-Rayleigh waves are of much lower amplitudes than the forward one and observed with a single transducer in pulse-echo mode. The objective of this paper is to report the observation of backward propagating leaky-Lamb waves in plates immersed in water. [Forward propagating leaky-Lamb waves were predicted and observed by Plona, Pitts, and Mayer, *J. Acoust. Soc. Am.* **59**, 1324 (1976).] Several symmetrical and antisymmetrical Lamb modes were identified in brass and stainless-steel plates by measuring the critical angle at which the back reflected signal (to the transmitter) is maximum. Backward propagating leaky-Lamb waves are also observed with a second transducer on the transmitting side of the plate. The measured dispersion curves are in good agreement with theoretical predictions. ^{a)} Also at Group de Physique des Solides, Université Paris, Paris 7, France.

2:50

RR6. Reflection and transmission of obliquely incident Rayleigh waves by a surface-breaking crack. Y. C. Angel and J. D. Achenbach (Department of Civil Engineering, The Technological Institute, Northwestern University, Evanston, IL 60201)

Reflection, transmission, and scattering of Rayleigh waves that are obliquely incident on a surface-breaking crack are investigated. The formulation of the problem has been reduced to two systems of singular integral equations of the first kind for the dislocation densities across the

crack faces. The systems of integral equations are solved numerically. Substitution of the dislocation densities into appropriate representation integrals yields the reflected and transmitted surface waves. Reflection and transmission coefficients are plotted versus the angle of incidence for various values of the wavelength, and versus the wavelength for various values of the angle of incidence. A critical angle of incidence, which depends on the material properties of the solid, has been observed. Beyond this angle no mechanical energy is radiated into the solid by body waves.

3:05

RR7. Eigenrays and eigenmodes for source-excited propagation in layered multiwave media. I. T. Lu, L. B. Felsen, and A. Kamel (Department of Electrical Engineering and Computer Science, Polytechnic Institute of New York, Route 110, Farmingdale, NY 11735)

Coupling of wave species at interfaces and boundaries in a medium composed of plane multiwave layers creates a proliferation of ray fields even after relatively few multiple reflections. This inhibits a ray treatment of propagation from source to observer. The difficulty may be overcome by diagonalizing, in a plane wave spectral representation of the Green's function, the reverberation matrix F descriptive of the boundary coupling. The resulting eigenvectors of F represent combinations of the original Q wave species, to be referred to as eigenrays, which, except for multiplication by the eigenvalue λ_q , $q = 1 \dots Q$, remain unaltered after one complete reverberation. Thus eigenrays may be traced through successive reverberations like ordinary rays in a *single-wave* medium. This feature also permits the original multiwave $Q \times Q$ matrix problem to be decoupled into a sequence of scalar problems. Conventional eigenmodes are generated from eigenrays by imposing self-consistency ($\lambda_q = 1$) after one reverberation. Alternative representations are given for the multiwave Green's function by use of these new concepts. [Work supported by National Science Foundation and ONR Acoustics Branch.]

3:20

RR8. Parametric acoustic conversion in ocean sediments. Suk Wang Yoon (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029) and Thomas A. Griffy (Department of Physics, The University of Texas at Austin, Austin, TX 78712)

The interaction of acoustic waves from a parametric acoustic array at a water-sediment interface has been studied theoretically. This investigation was motivated by the observation of better penetration of acoustical energy into sediments using a parametric acoustic array rather than a conventional source [T. G. Muir, C. W. Horton, Sr., and L. A. Thompson, *J. Sound Vib.* **64**, 539 (1979)]. In order to treat the nonlinear interaction of sound waves in a sediment, the inhomogeneous elastic wave equation for a solid was solved. One effect of the nonlinear terms in this equation is the coupling of compressional and shear waves. The separation of the coupling effect is shown. The primary waves from a parametric source are expected to excite Stoneley waves along the water-sediment interface. The Stoneley waves are assumed to serve as a source for parametric acoustic conversion in the sediment. The analytic solutions of the inhomogeneous wave equation are spherical waves modified by an angular dependence in the amplitude. [Work supported by ONR.]

3:35

RR9. Numerical simulation of acoustic propagation at a fluid-solid interface. Curt Randall (Schlumberger-Doll Research, Ridgefield, CT 06877)

Propagation of acoustic pulses in the neighborhood of the interface between fluid and solid half-spaces is investigated numerically using the finite difference technique. The solid half-space may be homogeneous, or it may comprise several distinct solids of differing acoustical properties. The solid-solid interface may have arbitrary orientation with respect to the fluid-solid interface. Plots of the displacement fields clearly illustrate the wavefronts, headwaves, and surface modes near the fluid-solid interface. In the case of the inhomogeneous solid, reflection, transmission, and mode conversions at the solid-solid interface, and subsequent interactions with the fluid-solid interface give rise to a multiplicity of decay pulses.

The origin and physical significance of each of these decay pulses is made evident by the displacement field plots.

3:50

RR10. Acoustic modes in a thin rigid cylinder with periodic gaps. S. K. Chang (Schlumberger-Doll Research, Box 307, Ridgefield, CT 06877)

Axisymmetric acoustic modes in a thin rigid cylinder with periodic gaps immersed in a fluid are studied. The gaps are circular slots on the circumference of the cylinder. The displacement potential in the fluid is expressed in terms of the Floquet waves in the cylindrical geometry. A Galerkin's method is applied to formulate the linear equations in relation to the boundary conditions on the cylinder and in the gaps. The modal wavenumbers are the location of the zeroes of the determinant of the matrix in the complex wavenumber plane. The zeroes are found by a Newton-Raphson iterative method. For an axisymmetric acoustic field, the modes are leaky waves. It was found that these leaky waves are fast waves that have a phase velocity faster than the sound waves in the ambient fluid medium. Since energy can leak out of the structure, these leaky waves are all attenuative in the axial direction. Longer gaps increases the modal attenuation since it is more difficult to trap the waves in the structure.

4:05

RR11. Scattering of acoustic waves by a periodic array of elastic strips. A. Sezginer and S. K. Chang (Schlumberger-Doll Research, Box 307, Ridgefield, CT 06877)

The scattering of a time harmonic plane wave in a fluid incident on an infinite periodic array of thin elastic strips is studied. The strips are coplanar and perpendicular to the plane of incidence. The acoustic displacement potential in the fluid is expressed as a Floquet wave expansion. Mindlin's theory of the bending of thin plates is applied to relate the pressure difference across the elastic strips to the motion of the strips. The shear and bending components of the strip motion are expanded in trigonometric basis functions, such that the shear force and bending moment are zero at the edges of the strips. The expansion coefficients are solved from the linear equations obtained by applying the method of weighted residuals to the boundary conditions on the strips and in the gaps. Anomalous reflection is observed when the incident wave frequency coincides with the transverse resonance frequencies of the strips. Resonance frequencies are observed in close pairs, corresponding to the strip resonant modes with a symmetric and an antisymmetric displacement profile. At normal

incidence, the antisymmetric family of modes are not excited. In addition, sharp changes in the Floquet wave coefficients occur, known as the Wood's anomalies, as the frequency of the angle of incidence is swept.

4:20

RR12. Acoustic waves along a fluid-filled borehole with a concentric solid layer. S. K. Chang and A. H. Everhart (Schlumberger-Doll Research, P. O. Box 307, Ridgefield, CT 06877)

Transient acoustic waves in a fluid-filled cylindrical borehole with a concentric elastic solid layer are complicated by acoustic modes that result from the presence of the solid layer. A real axis integration technique [Tsang and Rørdar, *Geophys.* 44, 1706-1720 (1979)] is used to compute waveforms excited by an isotropic point source placed on the axis of the hole. It is evident that dispersive acoustic modes are an important feature of the elastic wave propagation for this configuration. There are at least four classes of modes caused by the solid layer: the Stoneley mode, which is sometimes referred to as a tube wave; the compressional modes, caused by compressional wave reflections in the solid layer; the shear modes, caused by shear wave reflections; and the hybrid modes, due to mode conversions at the interfaces. A study of the dispersion, attenuation, and residue of the modes provides an explanation for most of the features in the waveforms.

4:35

RR13. Farfield decomposition of acoustic waveforms in a fluid-filled borehole. Andrew L. Kurkjian (Schlumberger-Doll Research, Box 307, Ridgefield, CT 06877)

In acoustic well logging, a transient acoustic source is placed within a mud-filled borehole for the purpose of exciting refracted compressional and shear waves in the surrounding formation. We model the downhole environment by an infinite fluid-filled cylinder surrounded by an infinite solid formation. An isotropic point source is placed on the axis of the hole and an exact solution for the field in the hole is obtained in the frequency-wavenumber domain. Space-time waveforms are obtained by numerically transforming this solution. The waveforms consist of compressional and shear refracted waves plus modes, which are guided by the hole. In this paper we numerically compute these individual components of the field. The modes are obtained by locating poles and evaluating residues. The refractions are obtained via numerical branchcut integrations. A vertical branchcut is used which converges rapidly but only gives accurate results in the farfield.

THURSDAY AFTERNOON, 10 NOVEMBER 1983

GARDEN ROOM, 1:30 TO 4:39 P.M.

Session SS. Speech Communication V: Phonemic Perception

John J. Ohala, Chairman

Department of Linguistics, University of California, Berkeley, California 94720

Chairman's Introduction—1:30

Contributed Papers

1:35

SS1. Perceived critical distances between $F1-F0$, $F2-F1$, $F3-F2$. Ann K. Syrdal and H. S. Gopal (Callier Center, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

A critical spectral distance of 3 to 3.5 critical bands (bark) between first and second formant frequencies in simplified two-formant synthetic

Russian vowels has been reported by Chistovich and her colleagues. Syrdal [J. Acoust. Soc. Am. Suppl. 1 71, S105 (1982)] transformed fundamental frequency and formant frequency values from naturally produced American English vowels into Bark values from which three distance measures were calculated: $F1-F0$, $F2-F1$, and $F3-F2$. Her proposed auditory-model of vowel perception reduced between-speaker variability and provided a perceptually based quantitatively defined link between some acoustic and phonetic features. In the present series of experiments, sever-

al synthetic two-vowel continua were constructed, each with a varying frequency parameter which crossed a critical distance in either of the 3-bark difference dimensions. Critical distances over the 3-bark difference dimensions, estimated by phoneme boundaries calculated from vowel identification tests, will be compared with each other and with previous estimates. Some effects of duration on the phoneme boundary estimates of critical distance will also be discussed. [Work supported by NIH].

1:47

SS2. A perceptually based approach to *F0* characterization. Philip Lieberman, Bill Katz, Roger Zimmerman, Allard Jongman, and Mark Miller (Department of Linguistics, Brown University, Providence, RI 02912)

While researchers applying quantitative techniques to the characterization of intonation patterns [W. E. Cooper and J. M. Sorensen, *Fundamental Frequency in Sentence Production* (Springer-Verlag, Berlin, 1981)] have claimed success in relating phenomena such as *F0* "resetting" to the presence of syntactic boundaries [J. Breckenridge and M. Y. Liberman, "The declination effect in perception," unpublished manuscript, Bell Laboratories (1977)], this procedure has been confined to a highly limited corpus of short, simple-declarative read sentences. The degree to which these models can be generalized to spontaneous speech has recently become the subject of controversy [N. Umeda, *J. Phonet.* 10, 279-290 (1982)]. We therefore examined a corpus of spontaneous and read speech with the intent of finding a quantitative characterization which could be applied across speaking conditions. To this end, we analyzed the data using the "topline" modeling procedure, as well as peak, valley, and all-points rms lines, computed by linear regression. In simple-declarative clauses and sentences, the "topline" model performed adequately. The all-points rms line also performed well and captured the greater degree of variance occurring in spontaneous speech. Given this result, and for reasons of computational simplicity, we felt that an all-points rms line was superior in characterizing the data. This rms model is consistent with auditory processing models.

1:59

SS3. On the mechanism of *F0* downdrift in Japanese. William J. Poser (Bell Laboratories, Murray Hill, NJ 07974 and Massachusetts Institute of Technology, Cambridge, MA 02139)

In Japanese, as in many other languages, fundamental frequency (*F0*) tends to fall during the course of an utterance. Previous studies attribute this fall to a single mechanism: accent reduction, downstep, or a declining tonal reference line. Data are presented showing that two mechanisms are involved. First, sentences containing only unaccented words were constructed in such a way as to permit perfect balancing of segmental content. Measurements of both average and peak *F0* on the high toned syllables of these words show a small but consistent decrease from left to right. The contour is approximately exponential, with a sharp downturn at the end. Second, the rate of fall is greater in sequences of accented nonsense words than in segmentally identical sequences of unaccented words. Similarly, the *F0* peak of a word of either type is lower following an accented word than following an unaccented word, in otherwise matched utterances. It is suggested that these facts are consequences of the simultaneous existence of downstep and declination in Japanese.

2:11

SS4. Fundamental frequency characteristics of southern Vietnamese phonemic tones in mono- and disyllables. Franz Seitz (Department of Linguistics, University of Pennsylvania, Philadelphia, PA 19104)

Fundamental frequency (*F0*) trajectories were measured for 87 Vietnamese monosyllables and 72 disyllables using a computer program which employs autoregressive spectral modeling with cepstrally based periodicity estimation. The study addresses the issue of invariance versus context-dependency of the *F0* patterns associated with the five phonemic tones; variability of pattern as a function of proximate tones, as well as segmental composition of the utterance, was examined. The findings are compared with those of the definitive acoustic phonetic studies of Vietna-

mese tones to date [M. S. Han, *U.S.C. Studies in the Phonology of Asian Languages VII* (1969); M. S. Han and K-O. Kim, *J. Phonet.* 2, 223-232 (1974)]. The present test utterances, unlike those of previous studies, were selected on the criterion of phonotactic representativeness. Han's canonical representations of the tones' *F0* patterns are corroborated in the case of monosyllables, with little effect from segmental composition. Disyllables involving sonorant consonants at the syllable juncture, however, do not have *F0* patterns that are sequences of two canonical trajectories. The *F0* pattern characteristics of these signals are discussed.

2:23

SS5. Measures of phonation types in a natural language. Paul L. Kirk (Department of Anthropology, California State University, Northridge, CA 91330), Peter Ladefoged, and Jenny Ladefoged (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

Using a Kay digital spectrograph, three types of analysis (spectrograms, power spectra, and waveforms) were used to investigate different phonation types in a single natural language (Jalapa Mazatec). In addition to normal voice, this language uses creaky voice and breathy voice to distinguish words. We recorded five speakers producing contrastive vowels that were identical in quality and pitch. The spectrograms showed that in comparison with normal vowels the creaky vowels had glottal pulses at irregular intervals and a greater amplitude in the higher frequencies. The breathy vowels are more clearly distinguished in the onset, and they often lacked discernible pulses. We used power spectra displays to quantify phonation type differences. In creaky vowels the amplitude of the first harmonic in relation to the amplitude of the first formant is less than in normal voice, and in breathy vowels the first formant had less intensity in comparison with that of the first harmonic. Waveforms displayed the irregularities in the glottal pulses and the high degree of first formant damping in creaky vowels. [Work supported by USPHS grant NS 18163-02.]

2:35

SS6. Atypical VOT categories in Hebrew and Spanish. Lawrence J. Raphael (Herbert H. Lehman College, CUNY, Bronx, NY 10468 and Haskins Laboratories, New Haven, CT 06510), Yishai Tobin (Ben Gurion University of the Negev, P. O. Box 653, Beer-Sheva, Israel 84105), and Tovah Most (Graduate School, CUNY, 33 West 42nd Street, New York, NY 10036)

Available data indicate that most languages employ three categories of Voice Onset Time (VOT) to distinguish homorganic stop consonants. The three categories, voicing lead/short lag/long lag, are temporally discrete and, in the opinion of some researchers, may coincide with psychoacoustically predetermined regions of auditory sensitivity. Moreover, speakers of languages which display a two-way opposition (i.e., /ptk/ vs /bdg/) seem to employ adjacent VOT categories (e.g., short lag and long lag) and not the extreme categories (i.e., voicing lead and long lag) to differentiate stops having the same place of articulation. Data gathered from native (but bilingual) speakers of Hebrew and Puerto Rican Spanish reveal that they commonly use either the extreme categories of VOT or voicing lead and a category intermediate to short and long lag. These data, together with those previously reported for Canadian French, Danish, and Korean, raise questions about the specific relationship between auditory sensitivity and the determination of VOT categories in natural languages. [Research supported by NINCDS.]

2:47

SS7. Perception of height differences in vowels. Peter F. Assmann and Terrance M. Nearey (Department of Linguistics, University of Alberta, Edmonton, AB T6G 2H1, Canada)

Vowels contrasting in the phonetic feature of height or openness are differentiated primarily by changes in the frequency of the first formant. Several hypotheses have been proposed to account for the perception of these differences. Mushnikov and Chistovich [*Sov. Phys. Acoust.* 19, 250-254 (1973)] argue that the frequency of the most prominent harmonic

in the *F1* region determines the height of a vowel. Carlson *et al.* [in *Auditory Analysis and Perception of Speech*, edited by G. Fant and M. A. A. Tatham (Academic, London, (1974))] favor an amplitude-weighted average (center of gravity) of the two most prominent harmonics. Results of matching experiments will be reported in which listeners select the best vowel match along an *F1* continuum corresponding to test stimuli with

modified amplitude spectra. The effects of *F0*, number, and frequency of harmonics are investigated. Predicted distances according to several models will be compared with listener's matches. Spectral distance models based on the excitation pattern or loudness density will also be considered. Results are generally consistent with a local center of gravity hypothesis. [Work supported by SSHRC.]

2:59-3:15

Break

Contributed Papers

3:15

SS8. Vowel allophones and the vowel-formant phonetic space. Patricia Keating, Marie Huffman, and Ellen Jackson (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024)

Through assimilation and other phonetic processes, a language may have more segments on the phonetic surface than in lexical representation. Consider English vowels: nasalized, long, front rounded, and central vowels are all introduced in derivations. Conceivably, then, languages could be much more similar phonetically than phonemically; possibly they could even fill the phonetic space equally. In that case, languages with fewer phonemes would have to show much more phonetic variation per phoneme, compared to languages with many phonemes. The alternative situation would be that languages with fewer phonemes would leave parts of the phonetic space unused. We are testing these possibilities by comparing the *F1-F2-F3* space for vowel allophones in different languages. Here we report on two five-vowel languages, Japanese and Russian. Seven speakers of each language read word lists and free text, and vowel formants are compared across conditions and languages. Preliminary results indicate that Japanese, the language expected to show the least variation, has widely separated vowels in word lists, but fills up the phonetic space in prose. [Work supported by USPHS grant NS 18163-02 to Peter Ladefoged.]

3:27

SS9. Problems with duration models: Evidence from Chinese. Ren Hongmo (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

Durations of (1) closure, (2) frication noise, (3) aspiration, and (4) voice in Chinese sounds under different linguistic conditions were measured on oscillogram displays of waveforms. A set of rules of percentage duration variation were formulated to reflect the effects of each of several linguistic factors such as distinctive features, syllable structures, word lengths, tone and stress patterns, etc. The notion of "incompressibility" [I. Lehiste, *J. Acoust. Soc. Am.* 51, 2018-2024 (1972); D. H. Klatt, *J. Acoust. Soc. Am.* 54, 1102-1104 (1973)], i.e., that durations resist being shortened under combined shortening rules, is discussed. Like English, Chinese vowel durations are incompressible under several shortening rules; unlike English, Chinese consonant durations do not show clear incompressibility. Two problems were found in applying the formula $D_i = K*(D_i - D_{min}) + D_{min}$ [D. H. Klatt, *J. Acoust. Soc. Am.* 59, 1208-1221 (1976)] in Chinese, one is that Chinese data show two values of "minimum duration" (one duration symptomatic of word length effects, and a shorter duration under neutral tone); another problem is that different combinations of shortening rules may result in different powers of incompressibility in duration.

3:39

SS10. An acoustic study of Japanese coronal fricative clusters resulting from vowel devoicing. Haruko Kawasaki (36-513 RLE, and Center for Cognitive Science, MIT, Cambridge, MA 02139)

Earlier studies [S. Shattuck-Hufnagel, V. Zue, and J. Bernstein, *J. Acoust. Soc. Am. Suppl.* 1 64, S92 (1978), V. Zue and S. Shattuck-Hufnagel, *Proc. 9th ICPS* 2, 215 (1979)] examining English fricative clusters [sʃ] and [ʃs] across word boundaries (e.g., in "gas shortage" and "tunafish sandwich") showed that the direction of assimilatory palatalization is predominantly anticipatory: [sʃ] often becomes [ʃʃ], but [ʃs] does not. This study investigates the cross-linguistic validity of such a tendency. Four speakers of standard Japanese casually spoke sentences containing fricative clusters [ʃs] and [sʃ] resulting from devoicing/deletion of an interven-

ing high vowel [i] or [u]. Preliminary observations reveal complex patterns not totally consistent with those for English: anticipatory palatalization in [sʃ] in Japanese does not seem as robust a process as in English. Furthermore, perseveratory palatalization, i.e., ʃs→ʃʃ, and both anticipatory and perseveratory depalatalization, i.e., ʃs→ss, sʃ→ss, all of which are relatively rare in English, are found in Japanese data. The results suggest a difficulty in explaining the previous English data in terms of peripheral articulatory control.

3:51

SS11. Vocalic formant transitions are integrated with noncontiguous fricative noises. D. H. Whalen (Haskins Laboratories, 270 Crown St., New Haven, CT 06510)

Natural-speech fricative noises from fricative-vowel or vowel-fricative syllables were edited so that the noise started out as either [s] or [ʃ] and ended up the other fricative. The overall duration of the noise was held constant, and the proportions of the two parts varied. The vocalic formant transitions supported the phonetic category of one part of the noise or the other. With both initial and final fricatives, the transitions reduced the amount of noise needed to identify a fricative as belonging to the category of the transitions. Thus either the transitions were integrated with noncontiguous acoustic cues, or the transitions had cue value only when contiguous with the appropriate fricative noise. A further experiment placed the same edited noises between two vocalic segments, and subjects identified two fricatives. The perceived ordering of the fricatives was often the reverse of the order of the noises themselves when the transitions were in the opposite order. It thus appears that the transitions are integrated with noncontiguous acoustic cues. [Work supported by NIH.]

4:03

SS12. Noise segment and vocalic cues of French fricatives. George P. McCasland (4045 Baltimore Avenue, C3, Philadelphia, PA 19104)

A fricative identification test was administered to native French speakers. Test stimuli had been edited from naturally spoken fricative-vowel utterances: /s, ʃ, f, θ/ - /i, a, u/. See [*J. Acoust. Soc. Am. Suppl.* 1 70, S33 (1981); and 73, S30 (1983)] for prior use with English and Spanish speakers. The results indicated that both noise segment intensity and spectrum were major acoustic cues for the French fricatives. These cues were redundant for the /t/, either low intensity noise or nonsibilant spectrum sufficed. A much weaker cue due to the original fricative, sibilant or nonsibilant, of the vocalic portion of the edited syllables must be taken into account. By comparison, Delattre had concluded using synthetic speech stimuli [*Phonetica* 18, 198-230 (1968)] that noise spectrum was necessary and noise intensity was redundant as cues for all French fricatives. Results implicate the specific language experience of the subjects, e.g., French lacks /θ/; the impact of natural or synthetic speech as sources of stimulus materials; and phonetic perception. Haskins Laboratories on NICHD Contract NIH-71-2420 was helpful in the preparation of stimulus materials.

4:15

SS13. Portuguese nasalized vowels and nasal consonants: Their relation to following consonants. Edith M. Maxwell (36-511 Speech Communication Group, Massachusetts Institute of Technology, Cambridge, MA 02139)

Pilot spectrographic results indicate that Brazilian Portuguese demonstrates a relationship similar to that found in English between syllabic nucleus duration and the voicing characteristic of a following obstruent:

vowels lengthen before voiced obstruents and shorten before voiceless obstruents. Also, a nasal consonant in Portuguese acts in conjunction with a preceding vowel as a syllabic nucleus; for example, if in a syllabic nucleus a vowel does not show the duration expected before a given obstruent, the nasal duration compensates, and vice versa. New data from six speakers of Brazilian Portuguese derived from computer analysis will be presented which confirm the preliminary results. The acoustic characteristics of nasalized vowels and nasal consonants in Portuguese will also be described, as well as the extent to which a prenasal vowel is nasalized. The difference between phonemically nasal vowels and vowels nasalized because they are adjacent to a nasal consonant (e.g., [lχ] lâ, "wool" vs [pãnu] pano, "cloth") will also be discussed.

4:27

SS14. Vowel-to-vowel coarticulation in Swahili. Rena A. Krakow and

Sharon Y. Manuel (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Acoustic data from Swahili (a Bantu language with a five-vowel system) reveal interesting differences in the extent and direction of vowel-to-vowel coarticulation as compared with other languages. Utterances were VpV disyllables, including all combinations of the five vowels with stress on the first vowel, recorded by a native speaker of Swahili. Two major results emerged from the analysis: (i) Previous research on other languages has shown vowel-to-vowel coarticulatory effects across the consonant to be restricted to the vocalic transitions. The present data reveal systematic vowel-to-vowel coarticulatory influence which extends into the steady-state portion of the transconsonantal vowel. (ii) A second important cross-linguistic difference is found in examining the relationship between the magnitude and direction of these effects. The first vowel, which is stressed, is found to be more affected by the quality of the second vowel than vice versa. In Swahili, then, anticipatory effects of coarticulation exceed carryover effects. This is in contrast to Fowler (1981) in which she shows stronger carryover than anticipatory effects for VCVs in English. [Work supported by NICHHD.]

THURSDAY AFTERNOON, 10 NOVEMBER 1983

CABINET ROOM, 2:00 TO 3:50 P.M.

Session TT. Shock and Vibration IV: Damping Techniques and Materials

Thomas S. Graham, Chairman

Bolt Beranek and Newman Inc., Union Station, New London, Connecticut 06320

Chairman's Introduction—2:00

Contributed Papers

2:05

TT1. **Transmission, reflection, and absorption of structureborne noise by discontinuities on a fluid-loaded plate.** D. Feit and J. Caspar (Department of Ship Acoustics, David Taylor Naval Ship R&D Center, Bethesda, MD 20084)

The propagation of structureborne noise over a fluid-loaded structure with a number of discrete structural discontinuities is modeled. An analysis is presented for an elastic plate excited by a uniform time harmonic line force; the structural discontinuities are each modeled as a translational (zero moment impedance) single degree of freedom (SDOF) vibrational system (mass-spring-dashpot) attached to the plate. Wave energy incident on the discontinuities are either reflected, transmitted absorbed by the acoustic medium in the form of radiated sound waves, or absorbed by the dashpot of the attached vibrational system. The reflection, transmission, and absorption coefficients are presented as a function of frequency for a finite number of periodically spaced discontinuities. The frequency dependence of these coefficients are affected by the resonance frequency of the SDOF as well as resonance frequencies associated with the spacing of the discontinuities.

2:20

TT2. **Structure-property relationships of some RIM polyurethanes for potential sound damping applications.** Rodger Capps, Linda M. Martin, and Eric D. Rustin (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856)

Polyurethane elastomers are block copolymers made up of relatively short urethane hard block segments separating longer flexible polyether or polyester soft blocks. Polyurethanes often exhibit mechanical properties superior to conventional elastomers. They also frequently display considerable damping. When combined with other types of semi-compati-

ble or incompatible polymers in polymeric networks known as interpenetrating network polymers (IPN's), they have the potential for high damping over a broad range of temperatures and frequencies. In an effort to understand how chemical composition and physical morphology affect the damping of these materials, a series of experiments involving varying weight ratios of hard block to soft block segments, molecular weight of the soft block, different chain extenders, and catalyst were carried out. The reaction-injection molded (RIM) urethanes were based upon poly(caprolactone) glycols, modified 4,4'-diphenyl methane diisocyanate (MDI), 1,4 butane diol or trimethyl propanol, and dibutyl tin dilaurate. The dynamic Young's modulus was measured by a resonance technique for each group of samples over a range of temperatures and frequencies. The observed viscoelastic properties were correlated with thermomechanical transitions observed in differential scanning calorimetry analysis. The potential of these materials for use in IPN's in sound damping applications is discussed.

2:35

TT3. **Comparative measurements on viscoelastic materials used for vibration damping.** Christian Pillot (Institut National des Sciences Appliquées de Lyon, 69621 Villeurbanne Cedex, France) and Geoffrey L. Wilson (Graduate Program in Acoustics and Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

In a previous paper at the 99th meeting of the Acoustical Society of America in Atlanta in 1980 a new viscoelastometer was described. This instrument enables the Young's modulus and loss factor to be easily obtained over a wide range of frequencies and temperatures. The prototype of this instrument is being used in a continuing project for the systematic evaluation of the properties of a wide variety of commercially available materials. The method will be described and some examples given.

TT4. Measurement of the damping characteristics of metal matrix composites using a resonant dwell technique. Nancy S. Timmerman (Bolt Beranek and Newman Inc., 10 Moulton Street, Cambridge, MA 02238)

The resonant dwell technique, first developed by Heine, was used to measure loss factors at the fundamental frequency of several cantilever beam samples. Nine different metal matrix composites and three unfilled base metals were tested to study stress and temperature dependence. Three temperatures and four stress levels were used in the study, and the fundamental frequency of all samples was between 100 and 200 Hz. The results indicate increasing loss factor with increasing stress level. Loss factors of the composite materials were lower at elevated temperature when compared with room temperature, while the loss factors of the unfilled base metals were higher. [Work supported by AMMRC.]

3:05

TT5. Theoretical prediction of sound attenuation in a liquid-filled rectangular flow duct using a viscoelastic-fluid composite liner. Sung H. Ko (Naval Underwater Systems Center, New London, CT 06320) and Louis T. Ho (Davis W. Taylor Naval Ship Research and Development Center, Annapolis, MD 20084)

A theoretical investigation was made of the sound attenuation in a two-dimensional, liquid-filled duct treated with a composite liner consisting of a viscoelastic slab and a fluid layer, which is backed by a perfectly rigid plate. The viscoelastic material is a rubber-like material that has a loss factor associated with the shear modulus, and is characterized by Lamé constants and the material density. The fluid layer of the composite liner is assumed to be lossless and nondispersive medium, and is characterized by the fluid density and the speed of sound. The fluid contained in the lined duct is assumed to be inviscid and characterized by the fluid density and the speed of the sound. The eigenvalue equation was derived based on the theory of elasticity, the acoustic wave equation in the presence of the flow in the duct, the acoustic wave equation in the absence of the flow for the fluid layer of the composite liner, and pertinent boundary conditions. The eigenvalue equation was solved numerically for a given duct geometry, the liner configuration, the composite liner material properties, and the flow velocity. Then, the sound attenuation was obtained using the calculated eigenvalues.

TT6. Narrow-band random response of a nonlinear oscillator. Huw G. Davies and Dennis Nandlall (Department of Mechanical Engineering, University of New Brunswick, Fredericton, N.B., Canada E3B 5A3)

The mean-square response of an oscillator with a nonlinear spring to narrow-band random excitation can exhibit multiple values in some frequency ranges. The randomly occurring jumps in level are similar to the jump phenomenon seen in the case of a swept sinusoidal excitation. The mean-square response of a Duffing oscillator is calculated here using equivalent linearization. The excitation is pink noise passed through a first-order resonant filter. Triple values of the response can occur at some frequencies only when the filter bandwidth is narrower than the oscillator bandwidth. The correct sinusoidal excitation limit is obtained as the filter bandwidth is decreased. On the other hand, the mean-square response is wholly single-valued if either the filter or oscillator bandwidths are large. The narrow-band mean-square response can be treated as a spectrum level. Integration of this level over all excitation frequencies shows good agreement with the exact mean-square response to white noise excitation. When multiple values occur the integrations using either the "upper" or "lower" part of the response spectrum bracket the exact value and can hence be used to give a crude estimate of the relative times the response is at the two pseudostable levels. Further work to estimate the expected time spent at each level is in progress. [Work supported by NSERC, Canada.]

3:35

TT7. A novel concept for vibration isolation. Thomas S. Graham (Bolt Beranek and Newman Inc., Union Station, New London, CT 06320)

This paper describes a novel concept for combining linear springs to synthesize various force-deflection curves as might be useful for particular problems of machinery isolation. (A force-deflection curve with a plateau is easily synthesized, for example.) The concept involves combining a negative spring, produced by a standard spring acting against lever arms, in series and parallel with standard springs. It is suggested that such devices may be useful where the use of rubber for machinery isolation is inappropriate.

Session UU. Noise VI: Industrial Hearing Conservation

Alice H. Suter, Chairman
Sonus Company, 1501 Red Oak Drive, Silver Spring, Maryland 20910

Chairman's Introduction—8:30

Invited Papers

8:35

UU1. Hearing conservation: Regulatory status and issues. Alice H. Suter (Sonus Co., 1501 Red Oak Drive, Silver Spring, MD 20910)

The Occupational Safety and Health Administration published the final decisions on its hearing conservation amendment on 8 March 1983. Certain changes are likely to cause confusion when OSHA personnel enforce the standard. Employers now must use personal monitoring in circumstances where area monitoring is "generally inappropriate" unless they can show that "area sampling produces equivalent results." Also, OSHA has changed the term "significant threshold shift" to "standard threshold shift," which some individuals interpret to mean an ordinary or expected shift in hearing level. Most changes from the January 1981 version are towards a performance rather than a specification approach, which may result in less rigorous hearing conservation practices, increasing the likelihood that employees will suffer noise-induced hearing loss before intervention can occur. Consequently, three critical issues have emerged. First, employers and consultants need a standard, practical method of testing hearing protector attenuation as it occurs in the field. Second, OSHA needs to derate the Noise Reduction Rating to reflect real world use, and third, employers need an objective method by which to evaluate the effectiveness of their hearing conservation programs.

9:05

UU2. Measuring the effectiveness of industrial hearing conservation programs. Julia Doswell Royster (Environmental Noise Consultants, Inc., P.O. Box 144, Cary, NC 27511-0144) and Larry H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27650)

The performance orientation of the hearing conservation amendment underscores the need to standardize methods for evaluating how adequately hearing conservation programs (HCPs) protect employees' hearing. The authors will review HCP evaluation procedures based on analysis of group audiometric data for exposed populations. Techniques include comparisons of mean thresholds for selected worker groups of hearing levels for nonindustrial noise exposed populations matched by race and sex, calculation of rates of change in mean thresholds, and analysis of threshold variability in the database toward better hearing or toward poorer hearing in baseline comparisons and sequential test comparisons. Acceptable ranges for the incidence of defined threshold shifts have been identified. Analysis results for additional industrial databases will be presented, as well as newly developed approaches to data analysis. Applying age corrections to individuals' thresholds before combining records into group data allows assessment of HCPs with fewer employees by pooling race/sex groups together. Year-to-year variability in thresholds for annual audiograms has been correlated with daily temporary threshold shifts for employees wearing different types of hearing protectors.

9:35

UU3. Measurement of employee noise exposure—Sound level meter procedures. Daniel R. Flynn (National Bureau of Standards, Washington, DC 20234)

Even in this era of sophisticated digital instrumentation, the simple hand-held sound level meter continues to hold its place as a workhorse in the measurement of employee noise exposure. The history of the sound level meter for such measurements is briefly reviewed and a summary is given of the major advantages and disadvantages of using a sound level meter for occupational noise exposure measurements. Some of the problems, perils, and pitfalls of conducting such measurements are highlighted. A brief review is given of accuracy requirements and calibration procedures. Spatial sampling problems are briefly discussed. The remainder of the talk concentrates on temporal sampling procedures for determination of employee noise exposure associated with time-varying noise; a simple procedure is described for accurate measurement of noise dose using a hand-held sound level meter.

10:05

UU4. Do dosimeters overestimate contributions from impulsive noise? John J. Earshen (Metrosonics, Inc., Box 23075, Rochester, NY 14692)

The Occupational Safety and Health Administration published the final version of its hearing conservation amendment on 8 March 1983. It states explicitly that contributions from all noise, impulsive, intermittent, and

continuous must be included in determining worker noise dose. In addition it is stated that measurements must be made with instruments possessing A-weighting and slow response. Many claims have been made that dosimeters overestimate contributions from impulsive noise (e.g., defense submissions and testimony in the Collier-Keyworth case and Friend of the Court submission by Chocolate Manufacturers Association). The CMA document presents results comparing theoretical computation for waveforms having short pulse components to results obtained with a variety of dosimeters currently in use. The instrument readings are high compared to the theoretical computations; thus it is concluded that dosimeters overestimate. This paper demonstrates that dosimeters complying with the OSHA stipulation to include A-weighting and slow response do not overestimate. Conclusions reached in the CMA brief are critically affected by ignoring the dynamic properties of Slow Response when performing theoretical determination of predicted dose for specified pulsed waveforms. It has been suggested, and in several instances demonstrated, that the discrepancy between computed and measured values disappear if fast response is substituted. Mathematical analysis of the idealized dosimeter transfer function verifies this conclusion. It should be noted that the discrepancy reappears for pulse duration of less than 0.125 s. The issue to be resolved is not one of instrument performance; rather it is whether the stipulation of A-weighted slow response is justifiable.

10:35

UU5. Assessment of the performance of hearing protectors for hearing conservation purposes. Elliott H. Berger (Acoustical Engineering, E-A-R Div., Cabot Corporation, 7911 Zionsville Road, Indianapolis, IN 46268)

Rating hearing protector performance for hearing conservation purposes can be most accurately accomplished when not only laboratory, but also field performance data are taken into consideration. Although standardized laboratory test data have been commonly available since the 1950's (ANSI Z24.22-1957; ANSI S3.19-1974), it is only since the mid 1970's that significant work has been conducted in the area of field performance evaluation of hearing protectors. Currently, a number of field techniques are available, including: real-ear attenuation at threshold and/or mid-line lateralization tests using actual noise exposed employees as subjects, either at their workplaces or at special test clinics; dosimetry studies via miniature microphones; and temporary threshold shift evaluations of hearing protection users in noisy industries. These real world methods will be reviewed and contrasted to the laboratory techniques, and key points illustrated with representative data.

Contributed Papers

11:05

UU6. Risk assessment for populations wearing hearing protectors. Felix Z. Sachs (U.S. Army Environmental Hygiene Agency, Bldg. E2100, Aberdeen Proving Ground, MD 21010)

The considerations and parameters underlying the assessment of risk of hearing loss for a population wearing personal hearing protective devices (HPD's) are examined. A damage risk function for a protected population is developed, and measures of hearing protector effectiveness are defined. It is noted that the Noise Reduction Rating (NRR) formulation of protector effectiveness cannot be validly used as a population-oriented measure of hearing protector effectiveness. In the population-oriented context, real-ear attenuation values for hearing protectors obtained under user-fit conditions must be used, but the mean value should only be reduced by 1/4 to 1/2 of a standard deviation in order to provide the required statistical confidence. Finally, some numerical examples of hearing protector effectiveness are developed using computer simulation akin to the Monte Carlo method.

11:20

UU7. Comparison of hearing protector single-number rating methods. Leonard C. Marraccini (U.S. Department of Labor, Mine Safety and Health Administration, Pittsburgh, PA 15213)

During the last ten years, various organizations have developed single-number rating methods for hearing protectors. These values have been used to approximate the amount of noise reduction obtained by using these devices. From the hearing protector attenuation data obtained from the American National Standards Institute test procedures, the organizations have mathematically arrived at single numbers such as the "R factor", the "NRR", and "D factor", and so on. These are based on various items such as actual noise spectrums, pink noise spectrums, standard deviation of the attenuation data, etc. This paper compares the various single number ratings which have been calculated for six different hearing protectors, and the methods used. As can be seen, the single number rating

varies considerably. Advantages and disadvantages of each method is also discussed. Finally, application of the values to on-the-job situations will be discussed.

11:35

UU8. The effect of reduced headband force on the attenuation of muff-type protectors. Daniel L. Johnson, Charles W. Nixon, and Mike Skelton (Air Force Aerospace Medical Research Laboratory, Wright-Patterson Air Force Base, OH 45433)

The headband force of five different protectors was decreased in one-half pound increments until the force was less than one pound. Using a dummy head, the attenuation was measured for each value of headband force. The results indicated that four of the muffs were relatively insensitive to a decrease in headband force and loss of 50% of the original force resulted in less than a 3-dB reduction in the value of the Noise Reduction Rating (NRR). One muff, however, was very sensitive and a 50% loss in headband force resulted in roughly a 50% loss in attenuation. To verify the dummy head results, attenuation was measured at selected forces on three human subjects. Although the attenuations measured were somewhat less than those measured for the dummy heads, the effect of changing headband force produced similar results. While all protectors were unaffected by a reduction of one-half pound in force, the effect of greater reductions varied dramatically with the make and model of the protector. Apparently, manufacturers need to set guidelines tailored to their specific models.

11:50

UU9. Impulse contribution of total worker noise dose. John Erdreich (National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, OH 45226)

For several years there has been controversy concerning whether impulsive noise should be included in determining the noise dose to which a worker is exposed. Clearly, the impulsive noise components reach the ear

and should be considered in determining hazard. However, recent court rulings held that, since appropriate instrumentation is not available to the industrial hygienist for impulse measurement, the calculations and tables in the OSHA Hearing Conservation Amendment should not include this noise. *How much of a difference does not including impulse noise make in the total dose?* Measurements in 14 plants made with a 40-kHz bandwidth

and either a 40-dB or a 72-dB dynamic range were analyzed and the total OSHA dose and the OSHA dose for the impulses alone were calculated. The impulses account for as much as 20% of the exposure time. In this short period, however, the impulse noise accounts for up to 50% of the total dose. Eliminating impulses from dose calculation may result in worker exposure estimates half of the actual dose.

FRIDAY MORNING, 11 NOVEMBER 1983

SENATE/COMMITTEE ROOMS, 9:00 A.M.

Session VV. Underwater Acoustics VI: Propagation (Précis-Poster Session)

Ding Lee, Chairman

Naval Underwater Systems Center, New London, Connecticut 06320

Chairman's Introduction—9:00

Contributed Papers

9:05

VV1. Theoretical method for a range-dependent fast field program. K. E. Gilbert, W. A. Kuperman, and R. E. Grimm (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL Station, MS 39529)

A fast field program (i.e., a Green's function method) is an accurate and flexible means for treating wave propagation in a range-independent environment. Until now, however, there apparently has been no theory for applying the method to a range-dependent environment. This paper derives and applies a fast field method for one-way wave propagation in a range-dependent environment. The method is based on the solution of the elliptic wave equation for an arbitrary vertical source distribution. On a given range step, the vertical source distribution is taken to be the field at the end of the previous step. It is shown analytically and numerically that this approach does, in fact, give the correct solution for one-way propagation in a range-dependent environment.

9:09

VV2. Matrix method for numerically solving the inhomogeneous wave equation. R. E. Grimm, K. E. Gilbert, and W. A. Kuperman (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL Station, MS 39529)

In recent years, matrix methods (e.g., finite elements, finite differences) have been successfully used to numerically solve the parabolic wave equation. This paper discusses the application of such methods to the solution of the elliptic wave equation for a general vertical source distribution. In particular, it is shown that one-dimensional finite elements are a convenient way to accurately reduce the depth-separated inhomogeneous wave equation to an easily solved system of linear equations. The advantages of this method over methods which use special functions are that it accepts arbitrary numerical profiles and that it can be simply applied to compute a range-dependent "fast field" solution. In addition, for realistic environments with complex stratification, the computation speed should be competitive with methods based on special functions.

9:13

VV3. A new fast-field solution to propagation in multilayered solid environments. Henrik Schmidt (SACLANT ASW Research Centre, 19026, La Spezia, Italy)

During the last decade the fast field program (FFP) has been an important tool for modeling wave propagation in stratified media. The earlier models of this type were based on a solution of the depth-separated wave equation by means of the Thomson-Haskell method. In this new FFP model a more direct and faster solution technique is employed. The field is expressed in terms of unknown scalar potentials. At each interface the boundary conditions yield a linear system of equations in the Hankel transforms of the potentials. Using a technique similar to the one used in finite-element programs, these equations are mapped into a global system which is solved with an efficient band-solver. Contrary to the Thomson-Haskell method, the new technique can treat several sources and receivers with one single solution. Hence, the field produced by vertical source arrays can be determined. Even with one source and receiver the computational speed has been improved by an order of magnitude, and pulse calculations by means of Fourier synthesis have become feasible. The features of the model are illustrated through a few computational examples.

9:17

VV4. Modeling of sound propagation over Dickins Seamount. Finn B. Jensen (SACLANT ASW Research Centre, 19026 La Spezia, Italy)

Experimental data for sound propagation across a steep seamount (14° mean slope) was recently reported [G. R. Ebbeson and R. G. Turner, *J. Acoust. Soc. Am.* **73**, 143-152 (1983)]. The broadband data exhibit a frequency-dependent shadowing effect with a drop in sound level just behind the mount of more than 20 dB at 12.5 Hz. An earlier attempt to explain these propagation features using the parabolic-equation technique gave promising results [F. B. Jensen, W. A. Kuperman, and H. Medwin, *J. Acoust. Soc. Am. Suppl.* **1 68**, S52 (1980)]. The data have now been subjected to a detailed modeling using both range-dependent ray theory and the recently developed wide-angled parabolic equation theory [D. Lee, G. Botseas, and J. S. Papadakis, *J. Acoust. Soc. Am.* **70**, 795-800 (1981)]. It is found that the full complexity of the propagation situation is well handled by both the acoustic models, and that the shadowing is due to the cutoff of steep propagation paths (bottom-bounce paths) by the seamount, leaving mainly waterborne paths to propagate beyond the mount. For these latter paths the surface-decoupling loss essentially determines the frequency dependence of the measured loss.

9:21

VV5. A range reflection parabolic equation. Frederick D. Tappert (University of Miami, Miami, FL 33149) and Ding Lee (Naval Underwater Systems Center, New London, CT 06320)

Application of the standard parabolic wave equation to solve real problems requires a clever selection of the reference wavenumber k_0 . An extended parabolic equation, earlier developed [F. D. Tappert, *The Parabolic Approximation Method* (Springer-Verlag, Berlin, 1977), Lecture Notes in Physics, Vol. 70, pp. 224–287], is re-introduced. This parabolic wave equation has range refraction capability and is totally k_0 -independent. The existing Implicit Finite-Difference (IFD) model was applied to test the range refraction parabolic equation. Results compare favorably with known solutions for weakly range dependent environments, but yield significant corrections for propagation through strong oceanic fronts. [Work supported by NUSC.]

9:25

VV6. HYPER: A hybrid parabolic equation-ray model for high-frequency underwater sound propagation. Frederick Tappert (University of Miami/RSMAS, Miami, FL 33149)

The parabolic equation (PE) propagation model contains all diffraction effects, PE applies to fully range-dependent environments, and PE has no upper limit of validity in frequency. However, simple operation count estimates show that PE execution times increase proportional to at least the square of frequency, and thus PE becomes impractical to run on minicomputers at high frequencies (above about 500 Hz). The equations of geometrical acoustics can be solved in fully range dependent environments with execution times independent of frequency, but amplitudes and phases are not reliable because these equations do not contain diffraction effects. The HYPER (Hybrid PE-Ray) model uses only the ray trace portion of geometrical acoustics to locate the important propagation paths. Then HYPER solves a modified parabolic equation in a small region (decreasing with increasing frequency) near the ray paths and computes amplitudes and phases. Theoretical calculations show that HYPER provides uniformly valid diffractive solutions in fully range dependent environments with execution times nearly independent of frequency. Thus HYPER should be practical to run on minicomputers at very high frequencies.

9:29

VV7. Propagation in a one-dimensional random medium. Suzanne T. McDaniel (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

Acoustic propagation in a one-dimensional random inhomogeneous medium is considered. Exact solutions of the Dyson equation for the coherent field are examined to determine constraints on the correlation function of the random inhomogeneities to insure finite results. The long range behavior of the coherent field is predicted, yielding an attenuation due to scattering and a modification of the real part of the propagation constant. Approximate solutions for the coherent field are then applied to determine the incoherent field. In contrast to results obtained using a straightforward perturbation approach, the predicted incoherent field is, in this case, finite in the absence of attenuation. [Research supported by ONR.]

9:33

VV8. A wide-angle three-dimensional parabolic wave equation. Ding Lee (Naval Underwater Systems Center), William L. Siegmann (Rensselaer Polytechnic Inst., Troy NY 12181), and Gregory A. Kriegsmann (Northwestern University, Evanston, IL 60201)

One straightforward extension of the standard two-dimensional PE to the three-dimensional case can be accomplished by retaining the angular derivative term. However, a number of basic questions arise concerning the derivation of this equation and its domain of validity. We discuss answers to these basic questions. Our answers lead to formulation of a new wide angle, three-dimensional partial differential equation which models sound propagation in the three-dimensional ocean. This formulation is achieved by operator theory. Instead of straightforward, small-angle three-dimensional PE, the new wide-angle equation is (in principle) a third-order partial differential equation. This third-order PDE is in an implicit form, making its solution by some numerical methods difficult.

However, we choose to solve our three-dimensional PE by the same Crank–Nicolson scheme as used to solve the two-dimensional PE. Using this implicit finite-difference scheme in conjunction with an efficient high-performance preconditioning technique, developed at Yale, can produce an accurate numerical solution. [Work supported jointly by ONR and NUSC.]

9:37

VV9. Application of the Yale sparse matrix technique to the three-dimensional parabolic wave equation. Ding Lee (Naval Underwater Systems Center, New London, CT 06320), Martin H. Schultz (Yale University, New Haven, CT 06520), and Kenneth R. Jackson (University of Toronto, Toronto, Ontario, Canada M5S 1A7)

The Yale University sparse matrix technique is an efficient method for solving large sparse systems of linear equations such as those that arise at each step in the numerical integration of the stiff system of ordinary differential equations resulting from the application of the method of lines to the three-dimensional parabolic wave equation. The theory behind the Yale sparse technique and the advantage of using it for this problem are discussed. The solution of a three-dimensional ocean eddy propagation problem is presented as an example. [Work supported jointly by ONR and NUSC.]

9:41

VV10. Time inversion of acoustic signals with weak dispersion. L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic Institute of New York, Route 110, Farmingdale, NY 11735) and E. Heyman (Department of Electrical Engineering, Tel-Aviv University, Tel-Aviv 69978, Israel)

Wave propagation in a vertically inhomogeneous ocean exhibits effective dispersion although the medium itself is nondispersive. Wave dispersion is weak at sufficiently high frequencies. The resulting local plane wave spectral representation of a source-excited sound field may then be inverted exactly to yield the impulsive response. Alternative inversion techniques are the Cagniard–DeHoop and the more general Chapman procedures. Both have limitations which are overcome by an analytic function approach, from which either a generalized Cagniard–DeHoop or the Chapman formulations are derived systematically by manipulations in the complex spectral plane. The three methods are compared and illustrated on caustic-forming multiple reflected ray fields and on refraction arrivals.

9:45

VV11. Intrinsic modes in a wedge-shaped ocean. J. M. Arnold (University of Nottingham, Nottingham, England) and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic Institute of New York, Route 110, Farmingdale, NY 11735)

By a recently developed theory applied to propagation in a wedge-shaped ocean with penetrable bottom [J. M. Arnold and L. B. Felsen, *J. Acoust. Soc. Am.* 73 1105 (1983)], spectral integrals have been derived that yield, by asymptotic evaluation to lowest order in the small bottom slope, the adiabatic modes, where these are valid, but which also describe the previously inaccessible transition through cutoff in upslope propagation. These spectral integrals have now been reexamined by performing the evaluation to the next asymptotic order. It is found that the result generated in this manner agrees, to the same order of accuracy, with a direct coupled mode solution of the boundary value problem wherein the *adiabatic* modes appear as the basic elements. Thus, the mode forms generated by the spectral integral include lowest order coupling between adiabatic modes. We refer to these uncoupled “more adaptable” modes, which likewise apply in the transition through cutoff, as *intrinsic* modes in order to distinguish them from the “less adaptable” adiabatic modes. Some consideration is given to still further generalization of the intrinsic modes to furnish uncoupled modes in a more strongly range dependent environment. [Work supported by ONR Acoustics Branch.]

VV12. Acoustic normal modes propagation through ocean eddies. Y. Desaubies, C. S. Chiu, and J. Miller (Department of Ocean Engineering, Woods Hole Oceanographic Institute, Woods Hole, MA 02543)

Uniformly valid asymptotic solutions are derived for normal mode propagation in a slowly varying ocean waveguide. The limits of validity of various approximations as a function of mode number, acoustic frequency, and range are discussed for realistic ocean environments. In particular, the first correction to the adiabatic equations is calculated, which provides an approximate solution to the coupled mode equations. The implications for numerical calculation of normal modes are discussed.

VV13. The uniform WKB normal mode code: Applications to cw and broadband propagation. R. F. Henrick (The Johns Hopkins University, Applied Physics Laboratory, Johns Hopkins Road, Laurel, MD 20707), J. R. Brannan, D. B. Warner, and G. P. Forney (Department of Mathematical Science, Clemson University, Clemson, SC 29631)

The increased interest in the modeling of underwater acoustic signals which are time limited or possess significant frequency bandwidth has resulted in a variety of broadband propagation codes. However, these codes are typically limited in utility by extremely long run times or in accuracy by simplifying assumptions. The approach taken here is to use uniform asymptotic approximations to the normal modes of the sound channel to obtain both accuracy and numerical efficiency. These approximations are based on the WKB approach but include corrections for modes that turn near boundaries and near mode turning points. This uniform WKB model approach leads to further efficiencies in multiple frequency broadband or pulsed propagation. The accuracy of this approach and the importance of these corrections is first illustrated for a cw signal. Subsequent examples show accurate results for both broadband and time pulsed signals. Run time comparisons show reductions of two orders of magnitude in comparison to conventional broadband codes.

VV14. On normal mode attenuation. Gary H. Brooke (Defence Research Establishment Pacific, Victoria, BC, Canada V0S 1B0)

The attenuation of normal modes due to viscous-type absorption losses in a layered bottom is examined. A standard perturbational approach is used to relate the modal attenuation coefficient to the energy stored and the power flow in each mode and to the intrinsic viscous loss in each layer. As such, the method is straightforwardly applied to geoaoustic waveguide configurations involving solid layers. Numerical examples are presented which illustrate the relationship between normal mode attenuation and energy concentration in the various layers.

VV15. Abstract withdrawn.

VV16. Impulsive wave propagation and the cuspid caustics. Michael G. Brown (Division of Ocean Engineering, Rosenstiel School of Marine and Atmospheric Science, University of Miami, Miami, FL 33149-1098)

The caustics of short wavelength wave propagation may be classified according to catastrophe theory. After establishing this connection the behavior of the time-dependent wave field due to an impulsive wave in the vicinity of the general cuspid catastrophe is examined. Numerical results are presented for the fold, cusp, and swallowtail catastrophes which show the smooth variation of the time-dependent acoustic field as the control parameters are varied. These results vividly demonstrate the phenomenon of ray focusing, showing the number of rays which converge at each catastrophe, the resulting phase shifts they undergo as well as their decay in shadow regions. Analytical results are given which describe the most important properties of the time-dependent acoustic field on and at large distances away from each of the cuspid catastrophes.

VV17. Impulsive wave propagation and the elliptic, hyperbolic, and parabolic umbilic caustics. Michael G. Brown (Division of Ocean Engineering, Rosenstiel School of Marine and Atmospheric Science, University of Miami, Miami, FL 33149-1098)

The time-dependent wave field due to an impulsive wave in the vicinity of the elliptic, hyperbolic, and parabolic umbilic caustics is examined. Numerical results are presented which show the smooth variation of the time-dependent acoustic field in the vicinity of all three caustics. Analytical results are given which describe the most important properties of these wave fields. The time-dependent structures of these wave fields are qualitatively different than those associated with the cuspid caustics. The reason for the difference stems from the difference in phase shifts which individual ray arrivals undergo as a result of touching one of the two types of caustic (cuspid versus umbilic).

VV18. Shallow-ocean fronts: Effects on array performance and acoustic prediction of properties. T. H. Rousseau,¹⁾ M. J. Jacobson, and W. L. Siegmann (Rensselaer Polytechnic Institute, Troy, NY 12181)

A parametric model is introduced which includes the location and orientation of a shallow-water front, as well as jumps in sound speed and current across it. In order to investigate frontal effects on array performance, a cw point source and a horizontal linear array of point receivers is considered. The error in predicted source direction, resulting from a frontally induced displacement in peak array output, is examined. Also, it is shown how the peak level and shape of the main-lobe array output are degraded significantly by the presence of a front as a point source approaches the nearfield. Finally, two sources and two receivers are placed at known locations on either side of the front. It is shown how travel times over direct rays between source-receiver pairs are affected by the various frontal parameters. A corresponding system of equations may be inverted so that travel-time changes can be used to predict estimates for frontal geometry and sound-speed and current discontinuities across a front. [Work supported by ONR.] ¹⁾Present address: Union College, Schenectady, NY 12308.

VV19. Long-range Pacific acoustic multipath identification. John Northrop and Richard C. Shockley (Naval Ocean Systems Center, San Diego, CA 92152)

Acoustic signals from three long-range (500-700 km) transmission paths in the Northeast Pacific were examined for multipath structure. Sound propagation along each path encountered both different sound-speed provinces and unique bathymetry which, together with the range differences, caused characteristic pulse arrival patterns at each hydrophone site. The receivers were all on a sloping bottom at depths between 1200 and 1400 m and the source depth was at 450 m in deep water. Ray-path arrivals were modeled using IMPULSE, a new ray-theoretical impulse response code written by one of us (RCS). This code uses piecewise continuous cubic polynomials to fit the sound-speed profile and bathymetric profile, and Runge-Kutta methods to solve the ray equation of motion. It allows arbitrary ray density in launch angle, and identifies eigenrays by searching for rays whose depths bracket the receiver and which were

adjacent at the source. Using this procedure, we were able to identify most of the major pulse arrivals observed at each of the recording sites.

10:21

VV20. Estimation of mesoscale sound speed perturbations via underwater acoustic tomography. Ching-Sang Chiu and Yves Desaubies (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

The inverse problem in ocean acoustic tomography corresponds to the estimation of the parameters characterizing the sound speed fluctuations due to mesoscale eddies from the acoustic travel time data. A deterministic model of the sound speed perturbation field δc is presented. In the model δc is assumed to be generated by a small number of Rossby waves and accounts for most of the data variance. Thus, δc is determined by only a few wave parameters, that is the amplitudes, wavenumbers and phases of the waves. The forward problem of travel time and ray path fluctuations due to δc and currents is studied by computer simulations and results are presented. An iterative method to estimate the wave parameters of the nonlinear system of the inverse problem is discussed and preliminary results in applying it to the data of the 1981 experiment are presented.

10:25

VV21. Sample effective sound velocity profiles. Edward R. Floyd (Naval Ocean Systems Center, Code 7222, San Diego, CA 92152)

Various sample *effective* sound velocity profiles [E. R. Floyd, J. Acoust. Soc. Am. **60**, 801–809 (1976)] for representative hypothetical sound velocity profiles are generated exactly albeit numerically. These various *effective* sound velocity profiles exhibit their dependence upon the constant of the motion (i.e., the vertex velocity), source frequency, and the source depth. Even for isotropic sound velocity profiles, the existence of a finite velocity gradient is sufficient to induce a constant-of-the-motion dependence in the *effective* sound velocity profile. Any dependence upon the constant of the motion represents an anisotropic *effective* sound velocity profile. All *effective* sound velocity profiles exhibit the vertex point receding to infinite depth (flat-earth assumption) which manifests the nodal characteristic of the singularity of the nonlinear differential equation that specifies the *effective* sound velocity profile. The behavior of the frequency dependence manifests an asymptotic approach to geometric

acoustics in the high-frequency limit. The source depth determines the initial values for the effective sound velocity profile and its derivative with respect to depth.

10:29

VV22. Measurements of transmission fluctuations in the Florida Straits. H. A. DeFerrari, D. Ko, and P. Gruber (Division of Ocean Engineering, Rosenstiel School of Marine and Atmospheric Science, University of Miami, Miami, FL 33149–1098)

Experiments are underway on the acoustic remote sensing of transport and average temperature of the Florida current. The experiments employ a reciprocal transmission geometry to separate transport and temperature effects. Preliminary results give information on the acoustic variability and coherence of transmission between a fixed bottom mounted source and moored receivers at ranges of 8, 16, and 24 km. Statistics of fluctuations for broadband pulse-like transmissions are presented. Environmental data are used as inputs to ray and P.E. models to interpret results. [Work supported by ONR and NOAA.]

10:33

VV23. Acoustic propagation of AFAR: Theory and experiment. Stephen A. Reynolds, Stanley M. Flatté (La Jolla Institute, P. O. Box 1434, La Jolla, CA 92038), Roger Dashen (Institute for Advanced Studies, Princeton, NJ), Barry Buehler, and Pat Maciejewski (NUSC, New London, CT)

We examine the statistics of the second and fourth moments of the acoustic field measured at the Azores Fixed Acoustic Range (AFAR). The results are from pulse transmissions during 1973 and 1975 of several acoustic frequencies over two volume-refracted paths in different scattering regimes: a 2.9-km geometric and a 35-km partially saturated path. The Garrett–Munk internal wave model is used to specify the index-of-refraction fluctuations; model parameters are entirely determined by environmental observations. Predictions based upon the path-integral formulation [*Sound Transmission through a Fluctuating Ocean*, edited by S. M. Flatté (1979)] compare favorably with the acoustic observations. The arrival time (phase) fluctuations are predominantly geometric. However at 35 km, microray effects in the phase can be identified. The 2.9-km amplitude behavior appears geometric although comparison is hampered by a short time series. At 35 km, log-amplitude and intensity fluctuations are due largely to microray effects; the power spectra of log-amplitude are described by a single parameter model where the parameter is related to the intensity decorrelation caused by the microrays. [Work supported by ONR, Code 425UA.]

FRIDAY MORNING, 11 NOVEMBER 1983

CHAMBER ROOM, 8:30 TO 11:50 A.M.

Session WW. Engineering Acoustics VI: Transducers

Daniel Queen, Chairman

Daniel Queen Associates, 5524 W. Gladys Avenue, Chicago, Illinois 60644

Chairman's Introduction—8:30

Contributed Papers

8:35

WW1. Reciprocity calibration of transducers in a plane-wave resonator. Charles L. Burmaster and S. L. Garrett (Physics Department, Code 61Gx, Naval Postgraduate School, Monterey, CA 93940)

In an article by Rudnick [J. Acoust. Soc. Am. **63**, 1923 (1978)] a procedure for the "reciprocity calibration of transducers in unconventional geometries" was described. The theory presented by Rudnick has been tested experimentally and extended to include the effects of the compliance of nonideal transducers on the quality factor of the plane-wave

resonances. A computer controlled system with automated switching was designed and built to acquire and process the resonance data required for the reciprocity calculations. Two reversible transducers and one microphone were mounted in the cylindrical resonant cavity to allow a six-way round robin self-consistency check of the experimental precision. The absolute accuracy of the sensitivity thus obtained is then compared with the results of a conventional reciprocity technique [ANSI S1.10-1966 (R1976)] and/or manufacturers specifications. [Work supported by ONR.]

8:50

WW2. Computerized measurement and analysis of sonar transducer equivalent circuit parameters. L. J. Skowronek, D. V. Conte, O. B. Wilson, and S. L. Garrett (Physics Department, Code 61Gx, Naval Postgraduate School, Monterey, CA 93940)

Measurements of the electrical and mechanical properties of transducers are of importance in both their design and service in their evaluation. Following the formalism of Hunt [*Electroacoustics* (Acoustical Society of America, New York, 1982)], these properties can be determined by accurate measurements of complex impedance or admittance made throughout the frequency range of interest. Manual collection of the necessary data is tedious and particularly error prone for transducers with sharp resonances. A computer-controlled system, using an H-P 85, will be described which automatically acquires and plots the electrical data and calculates the equivalent electrical circuit parameters, coupling constants, and efficiencies. Sample data will be presented for piezoelectric and magnetostrictive transducers. [Work supported by ONR and the NPS Foundation Research Program.]

9:05

WW3. A wide bandwidth constant beamwidth end-fire array. Beatriz Ugrinovich and Elmer L. Hixson (Department of Electrical Engineering, The University of Texas at Austin, Austin, TX 78712)

An end-fire array using five electret microphones and "bucket-brigade" analog delay lines are used to produce the desired beamwidth at 353 Hz. A half-scale model is implemented by adding two more microphones and using three redundant microphones to give the same beamwidth at 707 Hz. High-pass and low-pass filters are used in the respective array outputs which are summed to produce a constant beamwidth over the octave. The design process is repeated to produce a constant beamwidth 3-octave array. Predicted and measured characteristics will be presented.

9:20

WW4. An improved fiber optic lever transducer. Frank W. Cuomo (Department of Physics, University of Rhode Island, Kingston, RI 02881 and NUSC, Newport, RI 02841)

The concept of a bifurcated fiber optic transducer has been described and applied to the measurement of pressure and temperature. This approach has been used to design low-frequency hydrophones [F. W. Cuomo, *J. Acoust. Soc. Am.* **73**, 1848-1857 (1983)]. The sensitivity of this device is proportional to the total number of adjacent transmit/receive fiber pairs used in the bundle. For good sensitivity the transmit/receive fiber distribution must be maximized while the total number of fibers must be large. This paper discusses an improved bifurcated fiber optic transducer comprising one transmit fiber and two receive fibers, each receive fiber having a different core diameter. The three fibers are separated at one end and combined at the distal end in the vicinity of a miniature reflective surface sensitive to axial motion caused by minute pressure changes such that any displacement of the reflector from equilibrium will increase or decrease the illuminated areas of the two receive fibers and will generate a processed output signal proportional to this motion. The major relevant improvements over prior designs include a probe of minimal dimensions, a greatly improved sensitivity, and an output independent of variations at the input.

9:35

WW5. Noncontacting, acoustic vibration sensor. W. G. Richarz, P. A. Sullivan, and Tao Ma (Institute for Aerospace Studies, University of Toronto, 4925 Dufferin Street, Downsview, Ontario, M3H 5T6, Canada)

The measurement of vibration of lightweight structures requires that any loading due to a vibration transducer be eliminated. A noncontacting displacement sensor has been developed to assist in the understanding of the dynamics of lightweight air cushion vehicle skirts. A remote vibration sensor is required to monitor the unsteady motion of these membranelike structures. A low-cost acoustic displacement sensor has been developed. A piezoelectric driver aimed at the vibrating surface; the reflected beam is detected by a microphone. The motion of the reflector introduces a "Doppler shift" which is detected by a lock-in amplifier whose internal electronics transform this information into a signal proportional to displacement. The resolution of the system is governed by the wavelength of the incident beam, high frequencies ($f_0 < 20$ kHz) affording peak to peak values of the order of 10 mm; for larger excursions lower frequencies are used. The system is sensitive to changes in air temperature, the speed of sound figuring in the Doppler factor. In addition the directivity of the transducer plays a role, the resultant amplitude modulated signal affording additional sensitivity. The displacement transducer is readily configured from standard components found in many laboratories and is ideal for the detection of low-frequency displacements ($f < 100$ Hz). [Work supported by the Natural Science and Engineering Research Council and Transport Canada.]

9:50

WW6. Finite element method applied to characterization of piezoelectric ceramics. B. Tocquet, D. Boucher (GERDSM, 83140 Le Brusac, France), J. N. Decarpigny, J. C. Debus, and P. Tierce (ISEN, 3, rue F. Baës, 59046 Lille Cedex, France)

Finite element computation of piezoelectric transducers requires the knowledge of elastic, piezoelectric, and dielectric coefficients of ceramics. These constants are generally not available from the manufacturer, and even, variations up to 20% can be expected in the worse cases. The classical experimental method, proposed by Mason, is based on admittance measurements and a complete determination of all the materials properties requires five resonator geometries. An interesting method, based on an adjustment of first modes of ceramic rings with a finite element code, is described. It has three advantages over the Mason method. First, it does not require particular resonator geometries; second, the experimental method is summed up on a simple impedance measurement of the whole ceramics; and third, the adjustment is unaffected by local nonhomogeneity of materials. In this paper, the whole method is developed. Then experimental and numerical results are compared for several ceramics.

10:05

WW7. Computation of a piezoelectric deep-submergence transducer by three-dimensional finite element method. B. Tocquet, D. Boucher (GERDSM, 83140 Le Brusac, France), J. C. Debus, and P. Tierce (ISEN, 3, rue F. Baës, 59046 Lille Cedex, France)

The advantages of deep-submergence transducer are well known although practical solutions to solve high-pressure problems are very few. In this paper, such a system is studied. The radiating area of the transducer is a ring, on the inner face of which several length expander motors are fixed in a star shape. Every motor is composed of a ceramic stack and a tail mass. No mechanical connection is able to transmit exterior static or dynamic forces to the motors. This transducer has mainly two resonance frequencies which can be used: one is lower than the resonance frequency of the ring alone, the other is slightly higher than the motor resonance. To analyze this transducer, a finite element computer code (ATILA) has been developed. In water, the vicinity of the transducer is modeled with 3D fluid elements up to an exterior surface, the boundary reflection being damped through use of a suitable radiation impedance. So, in air eigenmodes, transmitting voltage response, directivity patterns, and electrical impedance can be computed. In this paper, the experimental results obtained with a deep-submergence transducer are compared with the theoretical ones.

10:20

WW8. Response of an edge-supported, annular, electret earphone. Ilene J. Busch-Vishniac (Department of Mechanical Engineering, University of Texas at Austin, Austin, TX 78712)

In many electret transducers, the diaphragm is held above a metalized backplate by circular ridges machined into the backplate. Such transducers may be thought of as a series of annular electret transducer segments, each of which has an edge-supported diaphragm. In the treatment presented here the behavior of an edge-supported, annular, electret earphone is studied analytically in a two-step procedure. First, the static diaphragm deflection is determined. A stability criterion is developed which physically corresponds to a limit in the maximum stable static deflection. Second, the diaphragm dynamic motion is determined and the resulting sound pressure in the ear canal calculated. Using this method the linear response and distortion response may be separately analyzed. Both tend to increase in level as the transducer approaches the point of static instability. Experimental results agree well with the predictions of the theory. [Work done in part at Bell Laboratories, Murray Hill, NJ 07974.]

10:35

WW9. Characteristics of a particle velocity hydrophone based on hot-film anemometry. Pieter S. Dubbelday (Naval Research Laboratory, Underwater Sound Reference Detachment, P. O. Box 8337, Orlando, FL 32856)

It is desirable to measure particle velocity in an acoustic field in addition to pressure. The heat transfer inherent in hot-film anemometry is mostly a function of particle motion. The relation between particle velocity and voltage output in a harmonic hydroacoustic field appears different from that in a nonperiodic flow (King's law). For horizontal motion, the output is proportional to the velocity squared, with a frequency twice that of the acoustic field. If the motion is vertical, it modulates the steady free-convection flow off the film, and a linear relation ensues with the same frequency as the field. Experimental results were obtained by means of a horizontally driven open trough for the first case and a vertical inertial calibrator for the second. It was attempted to correlate the output, expressed as a Nusselt number amplitude, to the relevant dimensionless numbers: particle displacement relative to sensor diameter, Stokes, Fourier, and Grashof numbers. Preliminary experiments show that an imposed bias flow improves the sensitivity and directional response of the sensor [Work supported by Office of Naval Research.]

10:50

WW10. An automated admittance measuring system for electroacoustic transducers. Clementina M. Ruggiero and T. A. Henriquez (Naval Research Laboratory, Underwater Sound Reference Detachment, Orlando, FL 32856)

A fully automated method of obtaining admittance loops of piezoelectric transducers has been designed to be utilized in conjunction with the H-P 4192A impedance analyzer. The algorithm determines the frequencies at maximum and minimum admittances; the series and parallel resonance frequencies; the resonance and antiresonance frequencies as well as the mechanical quality factor of the element under test. The collected data are plotted in the form of admittance loops with evenly spaced points. This program has been implemented successfully at the Underwater Sound Reference Detachment as a means to rapidly obtain the admittance information of various loaded and unloaded underwater transducers and transducer elements. [Work sponsored by NAVSEA.]

11:05

WW11. Four accelerometer structural vibration intensity measurements on beams. G. P. Carroll and J. A. Clark (Mechanical Engineering Department, Catholic University of America, Washington, DC 20064)

Two accelerometer measurement methods do not accurately measure structural vibration intensity near boundaries for the lowest beam modes because the shear and moment-related intensity components are no longer equal. It was shown by the authors [J. Acoust. Soc. Am. Suppl. 1 73, S33 (1983)] that this is a serious limitation since it is often by the lowest modes that power is transmitted through complex structures. In this talk, a four accelerometer spectral averaging measurement approach originally proposed by Verheij [J. Sound Vib. 70, 133-138 (1980)] will be reviewed. A method for conveniently implementing this approach with currently available two-channel FFT signal processors will be described and results from an experimental study of a beam excited transversely at one end and damped at the other end will be presented. The experimental results will also be compared with theoretical solutions and with results obtained independently by holographic interferometry. [Research supported by David Taylor Naval Ship Research and Development Center, Washington, DC.]

11:20

WW12. Relationship between reflection and transmission coefficients of a linear passive junction. L. J. Maga and G. Maidanik (David W. Taylor Naval Ship Research and Development Center, Bethesda, MD 20084)

The impulse response function of coupled, one-dimensional dynamic systems can be described in terms of the wavenumbers of the constituent dynamic systems together with the reflection and transmission coefficients at junctions which couple the dynamic systems. The junctions also define the terminal positions of the dynamic systems. Constructions of some simple but useful junctions are considered and the corresponding reflection and transmission coefficients that can be assigned to the junctions are derived. The characteristics of these coefficients and their relationship with each other are discussed.

11:35

WW13. An acoustic microscope designed for observing the vocal fold tissue. Yojiro Inoue, Yuki Kakita, and Minoru Hirano (Department of Otolaryngology, School of Medicine, Kurume University, Kurume 830, Japan)

This paper describes our newly designed scanning acoustic microscope (SAM). Our concern is to determine the mechanical properties of each histological component of the vocal folds in normal and varying pathological states. Generally speaking, the images obtained by means of an acoustic microscope not only present structure of the object to be investigated but also reflect its mechanical properties. Our present SAM has the following features. (1) The frequency of ultrasonic waves is variable between 100 and 200 MHz, causing the resolution between 5 and 10 μm . (2) Logarithmic and linear conversion of the image intensity are available. The image converted by logarithmic function can cover a wide range of mechanical properties. (3) Tilting mechanism is installed to the sample bed so as to adjust the angle of the specimen. (4) The images stored in the digital memory can be displayed on an ordinary video system. The structure and function of our new SAM system will be demonstrated together with some preliminary data of vocal fold tissue. [Work supported in part by Scientific Grant #57870103 by the Japanese Ministry of Education, Science and Culture.]

Session XX. Architectural Acoustics VII and Musical Acoustics VII: Organ Technology Today

David L. Klepper, Chairman

Klepper Marshall King Associates Ltd., 96 Haarlem Avenue, White Plains, New York 10603

Chairman's Introduction—9:00

Invited Papers

9:05

XX1. Comparing the relative advantages of pneumato- and electroacoustic sources for organ tone radiation in churches. Daniel W. Martin (Baldwin Piano & Organ Company, Cincinnati, OH 45202)

The variety which exists within church architectural design, in liturgical forms of worship, in the relative location of church organs and the worshippers, and even in the structures (or lack thereof) which organ builders have provided in the immediate vicinity of the organ tone radiators, suggests that there is probably more than one aesthetically acceptable means for radiating organ tone into worship space. Yet some adherents to the traditional maintain that there is only one acceptable means for radiating organ tone, i.e., pneumato-acoustic. This analysis assumes (1) that equivalent room acoustics and tone source locations are provided for either pneumato- or electroacoustic tone radiators; and (2) that electrical input signals are provided to electroacoustic tone radiators which include the transient, spectral and modulation contributions to the tone by the traditional air stream interaction with a pipe resonator. What acoustical advantages and disadvantages remain for the two different types of tone radiation means?

9:35

XX2. How computer and electroacoustic technology are used to improve musical performance in church electronic organs. Duane A. Kuhn (Classical Organ Division, Baldwin Piano & Organ Company, Cincinnati, OH 45202)

Recent advances in technology enable the electronic organ to approximate closely the musical resources of the pipe organ while increasing the control and flexibility of those resources. To create a true ensemble of organ sound a patented multiple octave generator tuning system provides multiple frequency sources on a given note and variance in sources between different octaves of the same chromatic note, all under master clock control for tuning stability. A tone radiation system consisting of an electroacoustic pipe array with fifteen individual audio channels provides tone radiation characteristics similar to those of a group of air-blown flue pipes. The use of a microprocessor controlled combination action increases instrument reliability while reducing its cost. New control offered to the organist by this noise-free combination action includes pedal stop-to-manual division registration, visual identification of individual stops in the crescendo pedal sequence, and a new type of division cancel.

10:05

Presentations by Panel of Experts*Participants*

Ronald F. Ellis

M. P. Moller, Inc., 403 N. Prospect St., Hagerstown, Maryland 21740

Jean-Louis Coignet, Tonal Director

Casavant Freres Lte., C. P. 38, St. Hyacinthe, Quebec, Canada J2S 7B2

Manuel J. Rosales

Rosales Organ Builders, Inc., 160 Glendale Blvd., Los Angeles, California 90026

Lawrence Phelps

Allen Organ Company, Macungie, Pennsylvania 18062

Session YY. Speech Communication VI: Effects of Age on Production and Perception

Patricia K. Kuhl, Chairman

Department of Speech and Hearing Science, University of Washington, Seattle, Washington 98195

Chairman's Introduction—8:30

Contributed Papers

8:35

YY1. The interaction of age and response condition in the use of acoustic cues to voicing in final stops. C. Wardrip-Fruin and Sharon Peach (Department of Communicative Disorders, California State University, Long Beach, CA 90840)

The developmental use of vowel duration, final transition, and voicing during closure as cues to voicing in final stop consonants was investigated, using subjects ages 3 to 6 years and adult. The stimuli were alterations of eight stop-vowel-stop words. The presentation of each stimulus item was response contingent. There was evidence that the adults responded more to duration cues than did the children. The response to spectral cues was more similar for the three age groups. A comparison of the response of these adults with response of adults in a previous study suggested that condition of response had a very significant effect on the direction and/or magnitude of response to specific treatments. In the earlier study the stimuli were presented at 750 ms. The response contingent presentation of the stimuli in this study seemed to result in a very different decision strategy on the part of the listeners. For example, adult subjects judged the same stimulus item voiced (67%) at 750 ms ISI but voiceless (62%) under the open interval condition. Listeners gave much more definite judgments under the fixed interval condition, judging many stimuli ambiguous in the open interval condition. Implications of these findings for the design and the interpretation of speech perception studies will be discussed.

8:47

YY2. Vowel normalization by 3-year-old children. Cathy A. Kubaska and Richard N. Aslin (Department of Psychology, Indiana University, Bloomington, IN 47405)

A two-alternative forced-choice procedure was used to assess vowel normalization in 3-year-olds. The stimuli were edited natural tokens of [æ] and [ʌ] produced by male and female adults and male and female children. Subjects were visually reinforced for making an appropriate pointing response to one of two televisions. Initial training of the pointing response consisted of pairing an adult male [æ] with one television and an adult male [ʌ] with the other. After reaching a 90% correct criterion on the training stimuli, subjects were presented with blocks of eight stimuli (four [æ]s and four [ʌ]s). The tokens of [ʌ] by the female adult and the children had formant values which were closer in frequency to the male adult's [æ]. For subjects in the constant vowel group, all the [æ]s were assigned to one pointing response, and all the [ʌ]s to the other. For subjects in the mixed vowel group, two [æ]s and two [ʌ]s were assigned to each pointing response. The constant vowel group was able to categorize the vowels appropriately, despite the differences in absolute frequency; the mixed vowel group performed near chance. These results provide empirical evidence for vowel normalization in young children.

8:59

YY3. The internal structure of vowel categories in infancy: Effects of stimulus "goodness." DiAnne L. Grieser and Patricia K. Kuhl (WJ-10, Child Development and Mental Retardation Center, University of Washington, Seattle, WA 98195)

We investigated whether perceived differences in stimulus goodness, that is, the degree to which a stimulus within a phoneme category is considered to be a "good instance" of the category, predicted differential responses to these stimuli by young infants. Two points were chosen in an $F1/F2$ coordinate vowel space: one judged to be a "good" representative of the /i/ vowel category, the other a "poor" representative, based on adult judgments. Then, variants were synthesized by manipulating the first two formants in both distance and direction from these two points. The resulting stimuli formed rings around the "good" /i/ and the "poor" /i/. Thirty-two 6-month-old infants were tested in a generalization task using a visually reinforced head-turn procedure. Half of the infants were tested using the "good" /i/ and its variants, and half using the "poor" /i/ and its variants. Results showed that generalization around the "good" /i/ was significantly greater than around the "poor" /i/. Moreover, infant response patterns correlated with adult subjective "goodness" judgments.

9:11

YY4. Children's identification of synthesized /r-w/ continua for adult and child speakers. Ralph N. Ohde (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232) and Donald J. Sharf (Speech and Hearing Sciences Program, University of Michigan, 1111 East Catherine Street, Ann Arbor, MI 48109)

In a recent study [D. J. Sharf and P. J. Benson, *J. Acoust. Soc. Am.* **71**, 1008-1015 (1982)], adults were found to reliably identify synthesized /r-w/ continua modeled after adult and child vocal tracts after one testing session. In order to determine the stability of children's perception of these continua, we tested ten 6-7-year-old subjects with normal articulation and hearing. After passing a criterion test of at least 90% correct identifications of the end-point stimuli, each child listened to randomizations of the stimuli from the adult and child continua on two separate days. The results showed no significant difference in category boundaries between the two testing sessions for both adult and child continua. The children correctly identified the three stimuli at each end of the adult and child continua at least at the 95% level and the category boundary occurred within the three middle stimuli. These findings suggest that children are at least as reliable as adults in their responses to variations in the $F2$ and $F3$ separation of glide onset frequencies even when the absolute frequency spectrum of stimuli differ by a factor of 1.5. [Work supported in part by NINCDS.]

9:23

YY5. Acoustic analysis of cry of SIDS infants. John M. Heinz, Rachel E. Stark, Rosemary Condino, Michele Hege (John F. Kennedy Institute, 707 N. Broadway, Baltimore, MD 21205), and Alfred Stein-schneider (National SIDS Institute, Atlanta, GA)

Spontaneous cry recordings were obtained for approximately 1700 infants born at the University of Maryland Hospital over a 2½-year period. Where possible, recordings were obtained during both the first and fourth weeks of life. Of those infants recorded, four later died of SIDS. For one, recordings had been obtained during both the first and fourth weeks, for one only during the first week, for one only during the fourth week, and one only at two later times (9 and 14 weeks). Thirty-second samples of

continuous, fully developed cry were selected for detailed analysis from each of the SIDS infants and from 50 age-matched controls. For each cry segment, measures of duration, intensity, and fundamental frequency were obtained and the presence or absence of a number of cry features was determined. Preliminary analysis of the data for the two SIDS infants recorded in week one indicates that a number of the measures group toward the tails of the distributions of these measures for the controls, while the data for the two SIDS infants recorded in week four do not show this pattern. Implications of these results and further findings will be discussed. [Work supported by NICHD and the Aaron Straus and Lillie Straus Foundation.]

9:35

YY6. Characteristics of reduplicated babbling in 6-8-month old infants. Rachel E. Stark and Jennifer Bond (John F. Kennedy Institute, 707 N. Broadway, Baltimore, MD 21205)

Reduplicated babbling has been defined as sequences of identical or nearly identical consonant-vowel syllables. This type of vocalization appears quite suddenly in the premeaningful utterances of infants at 6-9 months of age. In the present study, reduplicated babbling was examined at the time of its onset in four infants of 6-8 months. Computer-assisted spectrographic analysis was carried out and amplitude and pitch contours were obtained. Measurements were made of syllable and sequence duration. A specially designed transcription system was also employed in data analysis. It was found that: (1) the timing patterns in reduplicated babbling differed in specific ways from those of earlier vocalizations; (2) that a variety of vocants (vowel-like sounds) were produced within a babbled sequence; (3) that manner of articulation of consonantal (closant) sounds varied but place of articulation did not within a babbled sequence. The implications of these findings for development of speech motor skills will be discussed. [Work supported by NICHD.]

9:47

YY7. Development of vowel articulation in children. Corine Bickley (Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

The development of children's ability to produce high versus low vowels and front versus back vowels was examined in a longitudinal study.

Acoustic measurements were made of the vowels in spontaneously produced words with known referents. Vowels were grouped by target vowel into high, front (/i/, /u/), high back (/u/, /U/), and low (/a/, /s/). Evidence for control of vowel height was measured in terms of first formant frequency (F1) change over time. Vowel backing was measured by changes in second formant frequency (F2). Average F1 and F2 values and standard deviations of F1 and F2 were determined from harmonic amplitudes in narrow-band discrete Fourier transforms and from local maxima in LPC spectra. Over periods of 5 months, the average F1 for high vowels decreased while the average F1 for low vowels increased. Also, standard deviations for each vowel group decreased with age. Control of backing was seen primarily in increasing average F2 for front vowels. Standard deviations of F2 for both front and back vowels decreased with age. [Work supported by grants from the U. S. Department of Education and from NINCDS.]

9:59

YY8. Acoustic characteristics of stop consonants in the speech of children. Ralph N. Ohde (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232)

Although there are theoretical and practical reasons for studying the acoustic correlates of speech sounds in children, few studies have attempted to examine characteristics of their production which may directly vary as a function of place of articulation. Research on the spectral characteristics of stops in adults have shown that frequency and amplitude parameters vary systematically and somewhat invariantly with place of articulation. The purpose of this study was to determine the acoustic correlates of stop consonants in the speech of children. Six children between 8 and 9 years recorded five repetitions each of voiceless aspirated /p^h, t^h, k^h/ and voiced /b, d, g/ stops in combination with the vowels /i, e, u, o, a/. Spectral characteristics of burst onsets were determined according to LPC analyses. The findings showed that place of articulation systematically influenced the spectral characteristics of stop consonants. Average resonant peaks for labials were generally below 2 kHz, and for alveolars they were generally above 2500 Hz. Spectral characteristics of velars, on the other hand, were highly influenced by vowel context; the resonant peaks were high preceding high front vowels and low preceding back vowels. The findings will be discussed in terms of these acoustic correlates of place of articulation and their variability in children's speech production. [Work supported in part by NINCDS.]

FRIDAY MORNING, 11 NOVEMBER 1983

GARDEN ROOM, 10:15 A. M. TO 12:20 P. M.

Session ZZ. Speech Communication VII: Impaired Production and Perception

Sandra L Hamlet, Chairman

Department of Hearing and Speech Science, University of Maryland, College Park, Maryland 20742

Chairman's Introduction—10:15

Contributed Papers

10:20

ZZ1. Procedures which alter the intelligibility of voiced stop consonants for hearing impaired listeners. M. F. Dorman, J. M. Lindholm, and M. T. Hannley (Departments of Speech and Hearing Science and Psychology, Arizona State University, Tempe, AZ 85287)

To assess the hypothesis that spread of masking from F1 is responsible for poor identification of voiced stop consonants by hearing impaired listeners, /bdg/ were synthesized with bursts and formant transitions in

configurations with and without the first formant. Synthesis in the /eI/ environment allowed minimal acoustic differences to cue the contrasts. To maximize the probability of finding evidence of masking spread the stimuli were presented at levels both above and below the point of maximum intelligibility. At no point on the function was performance without the first formant better than performance with the first formant. To investigate other ways in which the intelligibility of /bd/ could be enhanced, we altered, in separate manipulations, the amplitude and duration at signal onset of the cues for place of articulation. Increasing the amplitude of the onset cues improved identification accuracy for /d/ but not for /b/ or

/g/. In contrast, increasing the duration of the onset cues improved identification accuracy for /g/ but impaired accuracy for /b/ and /d/. These results and others suggest that the beneficial effects of prolonging the onset spectrum for /g/ results not from better frequency resolution due to longer processing time, but from the cue of duration *per se*.

10:32

ZZ2. On improving articulation index predictions of speech sound identification performance by the hearing impaired. Chaslav V. Pavlovic (Department of Communicative Disorders, University of Mississippi, Oxford, MS 38677) and Gerald A. Studebaker (Department of Audiology and Speech Pathology, Memphis State University, Memphis, TN 38152)

Predictions of speech sound identification performance by hearing impaired individuals were made using an articulation index (AI). The AI scheme used was one that had earlier been found accurate for normal hearing listeners. It was expected that some sensorineural hearing impaired individuals would exhibit reduced suprathreshold speech processing capacity, and that therefore, the AI procedure would need to be modified to accommodate those effects. To this end we measured several psychoacoustical variables (critical bands, upward spread of masking, temporal masking, tuning curves, etc.) that might be incorporated into the AI formulas to account for the degraded performance. Results for each of the individuals tested will be discussed. [Work supported by NINCDS.]

10:44

ZZ3. Principal component amplitude compression. Diane K. Bustamante and Louis D. Braida (RLE, Rm. 36-749, MIT, Boston, MA 02139)

Multiband amplitude compression systems which process each frequency band independently of the others have yet to prove advantageous for sensorineural hearing impaired listeners over good linear amplification systems. Although audibility is improved, certain speech cues are consistently degraded by multiband compression. Acoustic analysis indicates that multiband systems whiten the short-term speech spectrum, reducing the spectral detail. Analysis of speech reception errors suggests that this degrades speech intelligibility. Spectral detail can be preserved by utilizing the correlation of speech levels in different frequency bands. Rather than treating each band independently, the compression in each channel is a function of the levels in all the channels. One implementation of such a system is based on a principal component analysis of short-term speech spectra. Compression of the first component, which corresponds roughly to overall level, equalizes the energy in consonants and vowels much as does a wideband compressor. Compression of the second component, which corresponds roughly to spectral tilt, acts as a variable frequency gain characteristic. The principal component system implementation and measurements of the properties of processed speech materials will be presented.

10:56

ZZ4. Second-language speech tests in a deaf patient with cochlear implant. Carl G. Müller and Cornelia M. Kemppel (Speech and Hearing Center, University of Denver, Denver, CO 80208)

Some reports of speech understanding through cochlear implants have been difficult to evaluate because test material in a foreign language was not comparable to the widely used and highly standardized speech material in English. Thus the data and their implications have not been widely accepted. To overcome this, we have provided patient "CK" from the Viennese Implant Project with extensive English language training in vocabulary, grammar, and composition with the goals of evaluating her process of second-language acquisition and of corroborating the significant Viennese results by using standardized English language tests. Tape-recorded material was presented solely through electrical connection to the speech processing electronics supplied by the Hochmair group (Vienna), thereby ruling out possible acoustical input. Stimulation was delivered to a single channel electrode through amplitude limiting, frequency shaping, and high-frequency pre-emphasis circuitry. We report on results

obtained from selected portions of the Minimal Auditory Capabilities (MAC) Test (including CID Everyday Sentences), the Diagnostic Rhyme Test, the California Consonant Test, and other less difficult tests designed to reduce the role of specific language features. Results support those of the Hochmairs on the efficacy of single channel implant systems for speech understanding. [Research supported in part by a Biomedical Research Support Grant (NIMH).]

11:08

ZZ5. A prototype hearing aid for acoustic coupling of telephone and hearing aids. Richard G. Stoker, Marilyn French-St. George, and Jonathan Brown (Program in Communication Disorders, The Pennsylvania State University, 110 Moore Building, University Park, PA 16802)

Recent trends in the telecommunications industry have resulted in an increased number of telephone receivers which are not compatible with the induction coils contained in many hearing aids. This situation has developed in spite of efforts by the Federal Government to legislate telephone/hearing aid compatibility. This project was designed to evaluate a prototype hearing aid with specialized frequency shaping capability. In short, we wished to discover if current technologies were capable of creating a hearing aid which could be effectively used by hearing aid wearers to understand speech transmitted over a telephone without the use of magnetic coupling. Ten subjects from three different hearing loss categories: precipitous high-frequency loss, sloping high-frequency loss, and relatively flat, severe loss, were tested. Subjects were evaluated using three types of frequency shaping: "typical hearing aid response," low-frequency compensation, and flat gain condition (up to 40 dB of linear amplification beyond that normally required for free-field communication). Subjects were required to respond to recordings of the NU-6 WORD LIST presented in quiet and with 45 dB and 55 dB SPL of competing background noise (multitalking noise). The results of this study suggest that certain classes of hearing impairment may benefit from specialized circuitry which provides frequency shaping to the telephone signals received at the hearing aid microphone. This technique has also been shown to even be effective in moderate amounts of background noise, allowing speculation that acoustic coupling of hearing aids and telephones may be more realistic than commonly thought.

11:20

ZZ6. An approach to telephone communication between deaf and hearing persons. Jared Bernstein, R. Becker, D. Bell, H. Murveit, F. Poza, and G. Stevens (Speech Research Program, SRI International, Menlo Park, CA 94025)

It may be possible for a deaf person and a hearing person to converse over the telephone using special equipment only at the deaf person's end of the connection. The deaf person drives a text-to-speech converter with an efficient keyboard system. The hearing person speaks sentences in a word-at-a-time fashion. The words are handled at the deaf person's end by a large vocabulary isolated-word recognition system that displays a sentence lattice (a sequence of sets of likely words for each word spoken). The deaf person then tries to find a sensible path through the sentence lattice. For such a system to be successful requires adequate performance in: (1) human text generation speed, (2) text-to-speech intelligibility, (3) human one-word-at-a-time speaking, (4) large vocabulary IWR, and (5) human disambiguation of sentence lattices. We report preliminary experimental results that bear on requirements (1), (2), (3), and (5); we outline an approach to (4), the implementation of which is simulated. Experimental studies of (1) and (5) are run on deaf subjects. [This material is based upon work supported by the National Science Foundation under Grant ECS-8023527.]

11:32

ZZ7. Identification of novel words and sentences using a tactile vocoder. B. J. Frost, P. L. Brooks (Department of Psychology, Queen's University, Kingston, Ontario, K7L 3N6, Canada), D. M. Gibson, and J. L. Mason (Department of Electrical Engineering, Queen's University, Kingston, Ontario, K7L 3N6, Canada)

The present experiments are part of an ongoing evaluation of a tactile vocoder that has been developed at Queen's University. The subject used in the experiments has normal hearing and had previously acquired a tactual vocabulary of 250 words. In all experiments the subject and reader sat in adjacent soundproof booths with a window providing visual contact when necessary. The first series of experiments examined the subject's ability to identify words that the subject had not received tactual experience with before. The subject was not aware of the possible words that might occur and a given word was only tested once. In the lipreading alone (LA) condition 39% of the words were identified correctly compared to 69% correct in the lipreading plus tactile vocoder (LV) condition. The ability of the subject to identify novel words with the tactile vocoder alone (VA) was surprising and encouraging. These results will be discussed in detail. Experiments testing the subject's ability to recognize open set sentences showed an increase in correct identification of over 100% when the LV condition was compared to the LA condition.

11:44

ZZ8. Identification of words and manner features of speech by two profoundly deaf teenagers using a tactile vocoder. P. L. Brooks, B. J. Frost (Department of Psychology, Queen's University, Kingston, Ontario, K7L 3N6, Canada), J. L. Mason, and D. M. Gibson (Department of Electrical Engineering, Queen's University, Kingston, Ontario, K7L 3N6, Canada)

Two prelingually profoundly deaf teenage girls were taught to identify "live-voice" English words while wearing a tactile vocoder. To ensure the subject's deafness, hearing aids were removed and earplugs and headphones carrying white noise were worn. The reader sat out of sight from the subject so lipreading information could not be obtained. Words were introduced two or three at a time until the subject reached criterion on a 50-word list. The two subjects acquired a tactual vocabulary of 50 words with 21 and 25 h of training, respectively. Immediately after these word identification tests, experiments were carried out to test the subject's ability to identify the "manner of articulation" features of speech. Subjects could identify whether CV's contained a nasal, liquid voiced stop, unvoiced stop, or unvoiced fricative with accuracies ranging from 75% to 100% correct.

11:56

ZZ9. Vowel production in aphasia. John H. Ryalls (Department of Linguistics, Brown University, Providence, RI 02912)

A group of five anterior and seven posterior aphasic patients were recorded for their vowel productions of the nine nondiphthongized vowels of American English [Peterson and Barney, *J. Acoust. Soc. Am.* **24**, 175-184 (1952)], and compared to the productions of a group of seven normal speakers. All phonemic substitutions were eliminated from the data base. A linear predictive coding program was used to extract the first and second formant frequencies for each of the five repetitions of the nine vowels. The vowel duration and the fundamental frequency of phonation were also measured. Statistical analyses revealed that the duration and the fundamental frequency for the aphasics were significantly different from the normal subjects. Although there were no significant differences in the formant frequency means, there were significantly larger standard deviations for the aphasic group. This greater variability was considered as evidence of a phonetic deficit on the part of the aphasic speakers, in the context of preserved phonemic organization. [Work supported by ADAMHA.]

12:08

ZZ10. Vowel formant characteristics in Huntington's disease. Thomas Murry and Michael Caligiuri (Veterans Administration Medical Center, San Diego, CA 92161)

The speech of Huntington's disease (HD) has been described with a variety of terms including slow, hesitant, and staccatolike, yet virtually no data have been presented to quantify these characteristics. Acoustic analysis of the vowels in HD speech may provide information about the nature of the speech output. In this study, two questions were posed. (1) Are the vowels of HD subjects different from normal speakers as determined by the formant cluster patterns? (2) Are HD speakers capable of reaching vowel formant targets when more complex speech utterances are utilized? Ten subjects with HD ranging in age from 26 to 58 years old were studied. Each subject was asked to produce four vowels in isolation and in CVC syllables in which the initial consonant was either /d/ or /g/ and the final consonant was always /d/. Measurements of F_1 and F_2 were made for all vowels from wideband spectrograms. The results indicate that the isovowel targets, especially for F_2 , for HD speakers, were different from normal F_2 values. When compared to the isovowels, the formants in the CVC context reflected centralization. The results suggest that formant trajectories for the vowels in HD speech fail to reach normal isovowel formant targets and that as the stimulus complexity increases, approximation to normal vowel formant frequencies decreases. [Work supported by the Veterans Administration.]

FRIDAY MORNING, 11 NOVEMBER 1983

CABINET ROOM, 9:00 TO 11:05 A.M.

Session AAA. Shock and Vibration V: Friction-Generated Noise

Paul J. Remington, Chairman

Bolt Beranek and Newman Inc., 10 Moulton Street, Cambridge, Massachusetts 02238

Chairman's Introduction—9:00

Invited Papers

9:05

AAA1. Disk brake noise in rail transit vehicles. Lynn L. Faulkner and Charles W. Rodman (Applied Dynamics and Acoustics Section, Battelle's Columbus Laboratories, Columbus, OH 43201)

A critical parameter in the mechanism of disk brake squeal is the manner in which the coefficient of friction changes as a function of the instantaneous relative velocity between the friction material and the brake rotor. Substitution of alternative materials for asbestos as a friction material constituent has produced squeal in some brake designs but not in others, leading to the conclusion that one or more parameters not associated with

friction material characteristics operates to control the squeal mechanism. The probable nature of these parameters is reviewed along with a description of experimental work carried out to investigate the cause of squeal. [Work Supported by US-DOT-TSC and Washington Metropolitan Area Transit Authority (WMATA).]

9:35

AAA2. Mechanisms for the generation of railroad wheel squeal. Nicholas R. Dixon and Paul J. Remington (Bolt Beranek and Newman Inc., Cambridge, MA 02138)

Railroad wheel squeal is associated with rail vehicles rounding curves of small radii, and it is regarded as one of the most intense and annoying noise mechanisms on rapid transit systems. This paper reviews a theory for the generation of wheel squeal and provides experimental data from laboratory and field measurements that are compared with the theoretical analysis. Squeal occurs when the railroad wheel slides laterally across the rail head, and the theoretical analysis shows that the friction forces generated by this sliding can cause squeal if the coefficient of friction decreases with increasing relative velocity between the wheel and rail. Laboratory measurements of the coefficient of friction (μ) versus the lateral creep of a railroad wheel are discussed that verify the decrease in μ with increasing wheel rail relative velocity. A series of field measurements on several rapid transit lines is described that determined the influence of the ratio of curve radius to truck wheelbase on the occurrence of squeal and provided good agreement with the theoretical prediction for the onset of wheel squeal. Methods of controlling squeal have involved special damped or resilient wheels. A recent survey of the loss factor data of specialized wheels, known to suppress squeal, indicates that less wheel damping is required than that predicted by the present theory. Reasons for this discrepancy are discussed.

10:05

AAA3. Vibration damping to control railroad wheel squeal. Francis Kirschner (The Soundcoat Company, 175 Pearl Street, Brooklyn, NY 11201)

The squeal noise radiated by rapid transit wheels on sharp turns is an important instance of friction generated noise. It is caused by the lateral slip-stick frictional force between the wheel and railhead. Noise control methods such as rail lubrication, hard faced rails, tuned wheel dampers, and resilient wheels have not been entirely satisfactory from the standpoint of either cost effectiveness or of safety. This paper will focus on several lightweight vibration damping treatments which have been economical and effective means of reducing squeal noise. Several ring and constrained layer damping treatments which make use of efficient viscoelastic damping materials have resulted from extensive development and laboratory testing programs. Treatments ranged from 1% to 2% of the total weight of wheels weighing 450-800 lb. Wayside noise reductions of 34 dB have been measured in recent field tests on rapid transit systems. These measurements have shown excellent correlation with vibration spectra measured in the laboratory and confirm the effectiveness of vibration damping in controlling friction generated noise of this type.

10:35

AAA4. In-service evaluation of damped subway wheels. Jerome E. Mann (Cambridge Collaborative, Inc., P. O. Box 74, Cambridge, MA 02142)

A 1-yr in-service evaluation of damped subway wheels is being done on the New York City Transit Authority system as part of the Urban Mass Transportation Administration's Urban Rail Noise Abatement Program. Four types of damped wheels are being evaluated to determine their effectiveness in reducing wheel squeal. Acoustic data are being collected throughout the 1-yr program to monitor the acoustical performance of the wheels. Data are also being collected to determine the cost of the wheels, including any additional maintenance and inspection. This presentation focuses on results obtained through the first 6 months of the 1-yr evaluation. [Work supported by the United States Department of Transportation.]

Session BBB. Physical Acoustics VI: General Topics III—Mostly Scattering

Allan D. Pierce, Chairman

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

Chairman's Introduction—8:00

Contributed Papers

8:05

BBB1. Variational method for prediction of acoustic radiation from vibrating bodies. Allan D. Pierce and Xiao-Feng Wu (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Integral equation formulations such as discussed by Chertock, Copley, Schenck, Meyer, Bell, Zinn, Stallybrass, and many others, typically have nonsymmetric kernels and are not easily rephrased as a variational principle. However, a symmetric kernel does result from the normal derivative of the Kirchhoff-Helmholtz integral, with the exterior point subsequently allowed to approach the surface, but the integrand then has unmanageable singularities. A method due to Maue and Stallybrass allows this integral equation to be recast into a usable form (involving tangential derivatives of the unknown surface pressure) and this version in turn leads to a variational principle that has considerable promise for systematic approximate solutions of radiation problems in the low to moderate ka regimes. Validity of this formulation is substantiated by derivation of known results for vibrating spheres and disks. Numerical results for other geometries are expected to yield higher accuracy than previously because one can incorporate one's best "physical insight" in initial choices for the class of trial functions for surface pressure. If computations now in progress are completed, the example of an oscillating airfoil will be discussed, with some expected clarification of the anomalous results of the Brooks' experiment.

8:17

BBB2. Acoustic scattering from absorbent wedges. Mehmet Marsan and Sabih I. Hayek (Department of Engineering Science and Mechanics and The Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

The prediction of the sound scattered by impedance covered wedges is obtained by use of dual integral equations. The impedance of each face of the wedge is modeled as a point reacting complex quantity which is independent of the other face. The solution was constructed as an angular spectrum to satisfy the boundary conditions and Sommerfeld radiation condition. The solution kernel was obtained exactly and is in terms of circular functions. The solution of the scattered pressure was then obtained for farfield and mid-range by use of asymptotic techniques. This solution is much simpler than the one developed by Russian scientists [for example, see G. D. Malyuzhinets, "The Radiation of Sound by Vibrating Boundaries of an Arbitrary Wedge, Parts 1 and 2," *Sov. Phys. Acoust.* **1**, 152-174, 240-248 (1955)] which was obtained by a method similar to Wiener-Hopf techniques. Thus it is easier to use highway noise applications because of its simplicity. The solution for the diffracted pressure exhibits clearly the role of the incident and reflected shadow boundaries and shows there is one minimum in the scattered field which depends on the two surface impedances. For backscattered pressure, the solution exhibits two minima. In all cases, the scattered pressure becomes negligible near the wedge surfaces. [Work supported by NAVSEA.]

8:29

BBB3. A new approach to resonance scattering in acoustics and elasticity. A. N. Norris (Exxon Research and Engineering Company, P. O. Box. 45, Linden, NJ 07036), G. A. Kriegsmann, and E. L. Reiss

(Department of Engineering Sciences and Applied Mathematics, Northwestern University, Evanston, IL 60201)

A new method is described for determining the scattered field from an obstacle in the resonant frequency regime. The method is based on the assumption that the density of the obstacle is either much greater or much less than the density of the surrounding acoustic fluid. When the frequency of the incident wave is not close to a resonance frequency of the obstacle in vacuo the reaction of the fluid on the obstacle is small. However, this reaction is important near a resonance frequency. We use the method of matched asymptotic expansions to obtain a uniform approximation for the scattered field, valid at all frequencies. It is shown that the resonant peak bandwidth is related to the radiation cross section, for simple modes. Multiple modes are also considered. The method is illustrated by examples.

8:41

BBB4. Comparison of physical acoustics and geometric theory of diffraction predictions for backscattering by impedance-covered wedgelike scatterers. D. A. Sachs (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02238)

Radar scattering studies have shown that scattering predictions based on physical acoustics are generally invalid away from specularly reflecting directions. In this paper, comparisons are made between the predictions of physical acoustics and the geometrical theory of diffraction for acoustic backscattering by impedance-covered wedgelike scatterers. Substantial differences are seen in the results at angles of incidence well off the specularly backscattering direction where diffraction by the wedge edge dominates the return. [Work sponsored by Naval Underwater Systems Center, New London, CT.]

8:53

BBB5. Acoustic resonance scattering by a cylindrical shell. D. Brill¹ and G. C. Gaunaurd (Naval Surface Weapons Center, R43, White Oak, Silver Spring, MD 20910)

We present a study of the resonance scattering process that takes place when plane acoustic waves are incident on elastic cylindrical shells in water. The analysis proceeds along the lines described by us earlier [D. Brill and G. Gaunaurd, *J. Acoust. Soc. Am.* **73**, 1448-1455 (1983)] for solid elastic cylinders, and extends it to (hollow) shells. For aluminum gas-filled cylindrical shells of various thicknesses, we construct the families of zeros in the complex $k_1 a$ ($\equiv x$) plane that give rise to the Rayleigh and the whispering gallery waves circumnavigating the shell. We have isolated the proper modal shell resonances from their modal backgrounds and we have generated monostatic plots of background-suppressed form functions (i.e., cross sections) that agree well with recent experimental observations made by a team in France. We have noticed that as the shells become thinner, fewer families at poles seem to offer contributions, until for very thin shells, only one type of circumnavigating surface wave seems to be possible. Finally, by means of the RST-pole diagrams, we have also studied the more attenuated (and less significant) contributions of the Franz or creeping waves to the overall resonance scattering process. Work continues. ¹D. Brill is with the U.S. Naval Academy, Annapolis, MD 21402.

BBB6. Excitation of anomalous resonances for acoustical scattering from fluid loaded solid elastic spheroids. M. F. Werby (Naval Ocean Research Development Activity, NSTL Station, MS 39529) and Roger H. Hackman (Naval Coastal Systems Center, Panama City, FL 32407)

Conventional theories of scattering from fluid-loaded elastic solids indicate that there is a threshold in KA (K is the wavenumber and A some characteristic dimension of the object) below which one cannot excite Rayleigh resonances and for which the scattering behavior is essentially rigid. However, a systematic investigation of acoustical scattering from elastic solid spheroids reveals that anomalous behavior occurs for scattering in an angular region off the axis of symmetry. It was found that this behavior is more pronounced for higher aspect ratio objects and that the anomalous form functions shift down in KA with increasing aspect ratio. Further investigation revealed that this behavior actually corresponded to the excitation of surface disturbances corresponding to standing waves following a spheroidal path. We thus refer to them as spheroidal resonances, which are similar in nature to Rayleigh type resonances. Such resonances can only occur for elongated objects above a certain aspect ratio and are most strongly excited at specific angles. These results will be presented.

9:17

BBB7. Rigid scattering from fluid-loaded objects for high aspect ratios. M. F. Werby (Naval Ocean Research and Development Activity, NSTL Station, MS 39529)

We perform a systematic study of acoustical scattering from rigid fluid-loaded spheroids with aspect ratios ranging from 10 to 30 using the coupled high-order T -matrix formulation developed for high-aspect ratio problems. The $kl/2$ range will be from 6 to 50, where k is the wavenumber and l is the length of the spheroid. Until recently it has not been possible to explore this region of interest. Of particular interest is the monostatic angular distributions. In this region there is a proliferation of nulls due to the variation of paths from different points of the scatterer to the point of detection. The origin of the first and subsequent nulls can be related to the aspect ratio of the object by an expression derived from classical wave arguments. Results will be presented.

9:29

BBB8. The isolation of physical mechanisms associated with acoustical scattering from submerged elastic spheroidal and spherical shells. M. F. Werby (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL Station, MS 39529) and Roger H. Hackman (Naval Coastal Systems Center, Panama City, FL 32407)

Acoustical scattering from submerged spheroidal and spherical elastic shells exhibits varied behavior over a broad range of KA (K is the wavenumber and A the largest dimension of the object). This behavior is complicated due to reflections from the inner surface (except for very thin shells) resulting in a very pronounced interference phenomenon between the internal and specular reflections as well as effects from the resonances. A general theory based on elastodynamics that accounts for all these mechanisms is perforce very difficult to employ computationally. In addition, some KA regions do not include all the mechanisms and thus a more simplified theory could be used. Thus, if one can develop a procedure that can account for the individual mechanisms separately considerable progress will be made in solving the overall problem. We propose to develop a theory based on an ordering procedure that will allow one to isolate the various effects. Several assumptions and schemes will be examined and compared to solutions of the exact problem.

9:41

BBB9. Angular spectrum characterization of acoustical transducers. Richard K. Johnson (IREX, 69 Spring Street, Ramsey, NJ 07446)

The transmitting and receiving responses of an acoustical transducer in a fluid are characterized using the angular spectrum of plane waves representation. Nearfield and farfield calibration procedures for deter-

mining the parameters of the representation are described. Some of these procedures have been implemented, and results are shown. A simplified description is developed for the case of a circularly symmetric transducer which permits a significant reduction in computation. Forward and backward propagation of the radiated acoustical field can be performed using Fourier transform methods. This characterization is very effective for analysis of scattering problems, particularly when they are expressed in the T -matrix formalism. An example is shown for normal incidence back-scattering from a layered viscoelastic half-space.

9:53

BBB10. Scattering of elastic surface waves by a semi-infinite fluid-filled crack. W. Möhring (Max-Planck-Institut für Strömungsforschung, Böttingerstr. 4-8, D 3400, Göttingen, Federal Republic Germany)

Wave momentum considerations lead to the conclusion that scattering problems involving a fluid-filled crack (normal stress and displacement continuous, vanishing tangential stress) should be in many respects simpler to solve than similar problems with a zero-stress crack. To illustrate this, an elastic half-space with a semi-infinite fluid-filled crack parallel to the free surface and extending to the left is considered. Incident surface waves, both from the left and from the right are assumed. The Wiener-Hopf technique is used to determine the reflected and transmitted surface waves for different frequencies and different values of Poisson's ratio. It is found that, in terms of the normal free-surface displacement, scattering is larger for waves incident from the left. The method can be extended to a semi-infinite fluid-filled crack in an elastic layer. The solution can also serve as a starting point in well-known efficient methods to calculate the scattering of elastic waves by a finite crack or by a finite sized connection between two elastic layers. Similar results for zero-stress cracks have apparently been obtained only in symmetrical configurations.

10:05

BBB11. Sound scattering and radiation from a thin rod in a viscous medium. Wen H. Lin and A. C. Raptis (Components Technology Division (308), Argonne National Laboratory, Argonne, IL 60439)

Sound waves incident on a thin elastic rod whose radius is smaller than the wavelength of the incident sound induce flexural and uniform compressional oscillations in the rod. These elastic oscillations again radiate sound waves into the fluid medium and affect the scattered waves. This synopsis presents theory of sound scattering by, and acousto-elastic vibrations of, a thin elastic unbound rod in a viscous fluid. The shear viscosity of the fluid was considered in the solutions to the boundary-value problems concerning the sound scattering and the elastic response of the rod. Results show that the scattered compressional waves consist of the rigid-rod scattering of compressional wave, a monopolar wave due to the uniform pulsating of the rod, and a dipolar wave due to the flexural vibration of the rod. The scattered viscous waves consist of the rigid-rod scattering of viscous wave and a dipolar wave due to the flexural vibration of the rod. Acoustic resonances occur when the effective inertia force of the rod balances the stiffness force of the rod. The fluid viscosity and the scattering of sound give rise to radiation damping for the rod vibrations and affect significantly the acoustic resonances.

10:17

BBB12. Selective observation of elastic body resonances by using long pulses to excite a ringing response in the body. S. K. Numrich, W. E. Howell, and H. Uberall⁹ (Naval Research Laboratory, Washington, DC 20375)

An elastic body submerged in water will ring when insonified by sound whose frequency is the same as one of the resonances of that body. Correspondence has been established between these normal mode resonances of the body and the individual circumferential waves predicted by creeping wave theory. Insonifying the target with a relatively long sinusoidal pulse results in a series of superimposed responses consisting of the specular reflection and a series of creeping waves arriving after successive circumnavigations of the body. Echoes, both analytically and experimentally

obtained, from spherical targets will be used to identify the properties of the resonance ringing and associate the superimposed echoes with the resonance poles that can be predicted theoretically.⁴¹ Also at The Catholic University of America, Washington, DC 20064.

10:29

BBB13. Effects due to elastic response of some simple bodies on coherently processed broadband echoes. C. F. Gaumond, S. K. Numrich, and L. R. Dragonette (Naval Research Laboratory, Washington, DC 20375)

As an approximation underwater targets are sometimes modeled as point scatterers. However, broadband echoes from elastic bodies of finite extent can be modulated by circumferential or internal elastic waves. This modulation causes the echoes to be temporally broadened and more widely distributed on a cross-ambiguity plot. The extent of these elastic effects on pulse width, coherence, and ambiguity will be shown for several cases of solid and hollow bodies.

10:41

BBB14. Mixed-mode acoustic glory: Model and experimental verification. Kevin L. Williams and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

Scattering from large elastic or fluid spheres is enhanced along the backward axis due to a weak focusing. The effect of diffraction on this axial focusing was shown [P. L. Marston and D. S. Langley, *J. Acoust. Soc. Am.* **73**, 1464–1475 (1983)] to remove an unphysical divergence of scattered pressure predicted by elementary ray acoustics. Our previous model of this "acoustic glory" of elastic spheres was complete only for transmitted waves within the sphere which were either all longitudinal (L) or all shear (S). These contributions were separated by measuring the backscattering of short tone bursts from a glass sphere in water having $ka \approx 457$. Measured peak-to-peak pressures were typically within 5% of predictions [P. L. Marston, K. L. Williams, and T. J. B. Hanson, *J. Acoust. Soc. Am.* **74**, 605–618 (1983)]. In the present research we complete the theory for the case of mixed-mode glories in which there is at least one L to S or S to L mode conversion due to reflection. A novel calculation shows the focal parameters are independent of permutations of the mode sequence; however, the contributions to the scattering depend on sequence. Measured mixed-mode pressures (from the aforementioned sphere) are within 5% of the predictions. [Work supported by ONR. Marston is an Alfred P. Sloan Research Fellow.]

10:53

BBB15. Ultrasound propagation and scattering in randomly distributed particles. João C. Machado (COPPE/UFRJ, Department of Biomedical Engineering, C. P. 68510, Rio de Janeiro, Brazil), Rubens A. Sigelmann, and Akira Ishimaru (Department of Electrical Engineering, University of Washington, Seattle, WA 98195)

This paper deals with experimental and theoretical investigations aimed at increasing the basic understanding of the interaction of acoustic waves with random media. The ranges of frequency are 0.85–3.0 MHz and 5.0–6.0 MHz. The random medium consists of a suspension of polystyrene spheres (mean diameter 0.589 mm, standard deviation 0.066 mm) in a solution of water and sugar. For different concentrations, experimental results for attenuation and pulse broadening for transmission and backscattering, and attenuation as a function of the receiving angle for transmission only are presented. In the range of 0.85–1.1 MHz, the total scattering cross section of the particles depends on the 13th power of frequency. This dependence is due to the role of shear waves inside the particles. The power spectrum density of a burst of sine wave backscattered from the random medium is measured. The experimental and theoretical results agree well. The complex propagation constant of the coherent wave in the medium with large concentrations of particles is measured. It deviates from those predicted for a low-density medium. For the theoretical calculations, the size distribution of the particles was considered. [Work supported by CNPq in Brazil and the U.S. Army Research Office and the Office of Naval Research.]

11:05

BBB16. Intensity fluctuations of waves propagating in a random medium. Alan R. Wenzel (Naval Ocean Research and Development Activity, Code 340, NSTL Station, MS 39529)

A theoretical analysis of the wave field radiated by a point source in a one-dimensional random medium is presented. The analysis is based on the quasi-Rytov method, and includes both multiple forward-scatter and multiple backscatter effects. An approximate expression for the mean-square fluctuating intensity is derived under conditions of a weakly dissipative medium. This expression shows that at low and intermediate frequencies the intensity fluctuations are a maximum at the source point, decreasing exponentially with distance from the source. These findings are in qualitative agreement with analytical and numerical results obtained by Kohler and Papanicolaou [*J. Math. Phys.* **15**, 2186–2197 (1974)]. At high frequencies, however, this expression shows that, under certain conditions, the intensity fluctuations reach a maximum at some positive distance from the source, decreasing monotonically with distance in either direction from this maximum point. These latter results are in qualitative agreement with observations of intensity fluctuations of optical waves propagating in the lower atmosphere. [Research supported by NORDA.]

11:17

BBB17. Scattering from gratings of high aspect ratio compliant tubes embedded in an elastomer. Ronald P. Radlinski and Robert S. Janus (Naval Underwater Systems Center, New London, CT 06320)

An improved formulation for normal incident plane wave scattering from gratings of compliant tubes embedded in a viscoelastic layer immersed in fluid is presented. Approximations to boundary conditions at the interfaces between the interstice and compliant elements are eliminated by treating the interstice as a viscoelastic plate which has the material properties of the polymer matrix material. The compliant tubes are treated as two structurally and acoustically coupled plates with end conditions which correspond to those for the vibrational modes of a highly eccentric elliptical shell. Conditions on normal displacement, normal stress, and tangential displacement relate the compliant plates and viscoelastic plate boundaries to the surrounding viscoelastic layers. Comparisons of the insertion loss for gratings in fluid and gratings in viscoelastic layers in fluid will be shown. Limitations of the analysis will be discussed and calculations will be compared with experimental data.

11:29

BBB18. Precision measurements of the reflection coefficient and of the acoustic impedance of materials in water filled tubes. E. J. Skudrzyk (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

The reflection coefficient and the acoustic impedance of materials can be measured in tubes either by monitoring the amplitude and phase of reflected sinusoidally modulated pulses, or by determining the resonance frequencies and the damping observed for a continuously excited standing wave. The pulse method has the advantage of isolating the errors caused by the tube constructional arrangement, while the standing wave method integrates these errors with the properties of the test probes. The tube itself affects the experimental results due to vibrations that travel in the wall and are coupled into the water, by additional tube vibrations that occur at its ends and by reflections caused by discontinuities and supports. Also, the transducers will be considered in detail and the effects of damping. In this paper, the behavior of the measuring tube will be analyzed and recommendations will be made for the design of trouble-free test tubes. [Sponsored by NAVSEA OSH and by the Office of Naval Research (Code 479).]

11:41

BBB19. Abstract withdrawn.

Session CCC. Psychological Acoustics VI: Hearing Impairment; Applied Acoustical Analysis

Donald D. Dirks, Chairman

Head and Neck Surgery, Medical Center, University of California, Los Angeles, California 90024

Chairman's Introduction—9:00

Contributed Papers

9:05

CCC1. Patterns of phoneme identification errors in cochlear and eighth nerve disorders. Maureen Hannley (Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85282)

The purpose of this study was to determine whether patterns of phoneme identification errors differ as a function of signal presentation level among listeners with cochlear and retrocochlear auditory disorder. An analysis of the speech intelligibility performance of 15 patients with confirmed eighth nerve disorder was conducted, using confusion matrices derived from responses to a monosyllabic word list. The same analysis was conducted on the responses of a group of 15 patients with cochlear disorders, matched to the retrocochlear group for age and audiometric configuration. The results indicated that: (1) vowel errors were more prevalent and varied directly with increasing stimulus presentation level in the retrocochlear group; and (2) consonant errors did not differ in type or relative frequency between the two groups, nor was there a level-dependent effect. These results are supported by closed-set vowel identification tests and thus do not appear to be an artifact of open set testing. Vowel errors may account for a major part of the speech "rollover" phenomenon typical of retrocochlear dysfunction.

9:20

CCC2. Hearing aid using an emphasized speech for hearing-impaired subjects. Hidenobu Harasaki and Shinji Ozawa (Faculty of Science and Technology, Keio University, 3-14-1, Hiyoshi, Kohoku-ku, Yokohama 223 Japan), Hiroshi Ono, and Toshisada Deguchi (Department of Educations, Tokyo Gakuji University, Koganei-shi, Tokyo, Japan)

A new hearing aid which emphasizes the phonetic features of monosyllables is suggested in this paper. In vowel parts of speech, we applied a digital filtering technique to a formant emphasis. The 101 order transversal filter emphasizes formant frequencies (+10 - +30 dB). This filter enabled us to reduce misrecognitions of /i/ and /e/, /u/ and /o/ for hearing-impaired children. To emphasize consonant parts, three emphasis methods are experimented and discussed. One is an emphasis of amplitudes in consonant part, the next is a duration expansion of nasal consonant, and the last is an insertion of silence between stop consonants and the following vowels. The combinations of these methods were also experimented. The results of these experiments showed a distinct improvement in the recognition ratio of monosyllables.

9:35

CCC3. Calculation of the external-ear directional filter function of the KEMAR manikin. R. E. Davis, S. Koshigoe, and A. Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

In order to assess the quantitative reliability of the calculation of sound diffraction by the external ear using the boundary-integral-equation method [S. Koshigoe and A. Tubis, *J. Acoust. Soc. Am. Suppl.* 1 73, S76 (1982)], we calculate the directional filter function for the KEMAR manikin external ear. The calculations are based on the assumptions of blocked meatus conditions and the embedding of the external ear in a plane rigid baffle. [Work supported by NSF.]

10:05

CCC4. Temporal processing in cochlear implants. Robert V. Shannon (Coleman Laboratory, HSE-863, University of California, San Francisco, CA 94143)

Psychophysical experiments were conducted to assess the temporal processing of patients with multichannel cochlear implants. Threshold and suprathreshold temporal integration were similar to normals, with integration time constants of 100-200 ms and 50-100 ms, respectively. Gap detection decreased from 60 ms near threshold to 1-2 ms for loud stimuli, similar to normals at high frequencies. Forward masking decreased as the delay of the signal from the masker was decreased, with a time course similar to or slightly longer than normals. All of these measures indicate that temporal integration and adaptation in implants are similar to normals, indicating an intact central mechanism. One major difference between implants and normals was in the sensitivity to details of the temporal waveform. Normals hear little or no difference as a function of phase within a complex waveform, but the same stimuli can cause large perceptual differences for implant patients. Implications for implant processor design are considered. [Work supported by NIH.]

10:20

CCC5. Electrical stimulation of the auditory nerve in man: Dynamic range as a function of stimulus duration. Mark White (863-HSE, University of California, San Francisco, CA 94143)

Psychophysical tests with implanted subjects indicate that responses to short duration electrical stimuli (similar in duration to transitional segments in speech) exhibit very narrow dynamic ranges, on the order of 6-10 dB; whereas responses to longer duration stimuli can exhibit considerably larger dynamic ranges, on the order of 17-40 dB. A specialized form of compression that compensates for this variation of the subject's dynamic range may be useful. A very significant amount of compression is essential because of the narrow dynamic range elicited by electrical stimuli in these subjects. We have investigated a set of "split-spectrum," multi-channel speech processor configurations. These speech processors have been designed to compress the wide dynamic range of the acoustic environment into the very narrow dynamic range of the implant recipient. A set of compression systems have been compared to determine which strategies offer the most effective control of stimulus amplitude. [Work supported by NIH and Hearing Research, Inc.]

10:35

CCC6. Gap detection in sensori-neural and simulated hearing impairments. Mary Florentine (133FR, Communication Research Laboratory, Northeastern University, Boston, MA 02115) and Søren Buus (413MU, Auditory Perception Laboratory, Northeastern University, Boston, MA 02115)

The minimum detectable gap MDG in a low-pass ($f_c = 7$ kHz) noise was measured monaurally as a function of level in listeners with normal

hearing, hearing impairments of primarily cochlear origin, and impairments simulated by masking. The listener judged which of two intervals, marked by lights, contained the gap and a modified BUDTIF procedure was used to measure the MDG. The results for normal listeners agree with previous data. Results for seven impaired listeners indicate that in most cases their MDGs depend on their high-frequency thresholds. For example, at high levels, one listener with a low-frequency impairment had nearly normal MDGs, whereas listeners with high-frequency impairments showed enlarged MDGs. Most of the reduction in temporal acuity could be produced by presenting a normal listener with a masking noise spectrally shaped to simulate the audiogram. However, for two impaired listeners with almost identical audiograms, one listener's performance could be simulated well by masking a normal ear, but the other's MDGs were worse than the simulation by a factor 1.5 at the high levels. These results indicate that temporal acuity, *per se*, may be reduced in some, but not all, impaired listeners. [Work supported by NIH Grant RR07143 and the RSDf of Northeastern University.]

10:50

CCC7. Diffraction limitations on sound-imaging experiments for hearing-aided listeners. George F. Kuhn (Vibrasound Research Corp., 10957 East Bethany Drive, Suite J, Aurora, CO 80014)

It had been proposed by Bauer [J. Acoust. Soc. Am. 33, 1536-1539 (1961)] that two loudspeakers of different source strengths in the horizontal plane could be used to produce a virtual sound image, located at other than the actual source location. It is shown here that, due to the diffraction of sound by the head, this particular method produces simultaneous and interdependent interaural time (ITD) and level differences (ILD). The particular ITDs and ILDs produced by Bauer's method do not generally correspond to a unique spatial location and therefore produce potentially conflicting localization cues. This difficulty is aggravated by unbalancing the hearing aid gains or their phase response. It appears from the study by A. Nabelek, T. Letowski, and D. Mason [J. Speech Hear. Res. 23, 670-687 (1980)] that aided listeners position the sound image on the basis of interaural time differences. The role of interaural pressure level differences on the positioning of sound images could not be established unambiguously from these data.

11:05

CCC8. Masking of the /da-/ga/ distinction in cochlear impairment. Søren Buus, Joanne L. Miller, and Bertram Scharf (Psychology Department, Northeastern University, Boston, MA 02115)

Discrimination between two 300-ms synthesized syllables, /da/ and /ga/, presented in the middle of a 900-ms narrow-band noise, was measured in three normal and six cochlearly impaired listeners. The segmental distinction was conveyed by the frequency of the initial burst and the form of the third-formant transition, which was 50 ms in duration: For /ga/ the transition started around 1.9 kHz and rose to its steady-state value of 2.5 kHz and for /da/ it started around 3.0 kHz and fell to 2.5 kHz. The narrow-band noises were 300 Hz wide and were centered between 1

and 3.5 kHz. For the normal listeners, the speech was set to 70 dB SPL and the noise to 80 dB SPL. For the impaired listeners, the levels were chosen to yield better than 95% discrimination in the easiest condition, but poor discrimination in the most difficult condition. Both groups of listeners showed the poorest performance when the noise was centered at 2.5 kHz. As the noise moved away from this critical region performance improved for both groups, but more rapidly for the normal listeners. These results indicate that the impaired listeners' reduced frequency selectivity may contribute to their difficulty of discriminating speech in noise. [Supported by NSF and NIH.]

11:20

CCC9. Changes of tinnitus pitch after monaural exposure to a 1000-Hz pure tone. I. M. Young and L. D. Lowry (Department of Otolaryngology, Jefferson Medical College of Thomas Jefferson University, Philadelphia, PA 19107)

A subject with tinnitus of 10 000-Hz pitch equivalent in the left ear and no tinnitus in the right ear was exposed in the left ear to a continuous pure tone 2000 Hz, 107 dB SPL, for 10 min. This exposure resulted in permanent tinnitus in both ears with similar pitch of 10 000 Hz. When the left ear was exposed to a continuous pure tone 1000 Hz, 123 dB SPL, for 21 min, tinnitus disappeared temporarily from stimulated ear but was heard in the nonstimulated ear. In the left ear: 6½ h after stimulation, tinnitus reappeared as a mixture of multiple pitches; 51 h after stimulation, tinnitus pitch changed to 4100 Hz, 3 days later to 5800 Hz, 5 days later to 8700 Hz, and 48 days later to the original 10 000 Hz and has remained there since. In the nonstimulated right ear: After stimulation, tinnitus pitch was 7500-8500 Hz for 7 days; thereafter, tinnitus has been fluctuating between 8500-9500 Hz; tinnitus did not return to the pre-exposure pitch of 10 000 Hz until about 9 weeks after exposure. Difference in recovery between two ears was compared with our previous studies of 125-, 250-, and 500-Hz stimulation.

11:35

CCC10. Diagnosis of childhood respiratory disease based on acoustical analysis. Elizabeth B. Slawinska (Department of Psychology, University of Calgary, Calgary, Alberta, T2N 1N4, Canada)

Diseases of the respiratory pathways are often associated with a narrowing of the respiratory channel. Different diseases are characterized by a well defined locus of the constriction. Theoretical analyses suggest particular acoustical characteristics which can aid in diagnosis. Predictions include (a) a shift in spectral density to the higher frequency range, due jointly to an enlarged noise source and the constricted respiratory channel, in tonsillitis; (b) specific relations between the acoustical pressure/murmur at inspiration and expiration, in congenital stridor; (c) several noise sources generating large high-frequency amplitudes, in acute subglottal laryngitis; (d) stridor during inspiration if the constriction is in the neck or during expiration otherwise, with the narrowing of the traches. These theoretical predictions are developed and then tested with experimental data obtained from 500 children, including 50 cases of tonsillitis, 47 cases of congenital stridor, 30 cases of subglottal laryngitis, and 24 cases of narrowing of the traches. [Work supported by AHFMR and NSERC.]

Session DDD. Noise VII: Measurement, Evaluation, and Control

George C. Maling, Jr., Chairman

IBM Acoustics Laboratory, C18/704, P. O. Box 390, Poughkeepsie, New York 12602

Chairman's Introduction—2:00

Contributed Papers

2:05

DDD1. A note on the acoustics of a duct with a wall treatment of axially partitioned porous material. S. W. Rienstra (National Aerospace Laboratory NLR, P. O. Box 153, 8300 AD Emmeloord, The Netherlands)

Considered is the problem of sound propagation through a cylindrical duct with a wall treatment consisting of a layer of porous material, usually fixed by a segmented structure. The mathematical problem is one of sound fields in the duct and in the layer coupled by conditions of continuity across the interface. So, in general, the layer is not locally reacting and cannot be represented by an impedance of the wall. As a result calculations tend to be quite complex. However, in the present paper it is shown that if the porous layer is segmented by an axial array of annular partitions with a small enough pitch, the coupling of the fields simplifies in such a way that the condition of continuity reduces to a boundary condition, *per circumferential mode* similar to that of a point reacting liner. So then the porous material is characterized by a circumferential mode number-dependent impedance of the duct wall. Consequently, all the well-established theory for sound propagation in ducts with uniform wall impedance is applicable from here on.

2:20

DDD2. The effect of grazing flow on the impedance of Helmholtz resonator. Nghiem-Minh Nguyen-Vo, and Peter A. Monkewitz (Mechanics and Structure Department, University of California, Los Angeles, CA 90024)

A theoretical model for the acoustic impedance of a Helmholtz resonator with grazing flow is presented for the case of low-excitation frequency and amplitude and moderate grazing flow Mach number. The analysis is based on the two-dimensional linearized inviscid equation of motion. The method of matched asymptotic expansions is combined with the Green's function technique to solve the boundary value problem. It is shown that the vorticity generated by sound incident on a rigid body is "carried away" by the grazing flow. This results in a transfer of acoustic energy to the one associated with vortical distribution. The acoustic impedance obtained agrees well with experimental results available in the literature. [Research supported by NASA.]

2:35

DDD3. Acoustic fatigue failure of aircraft component. Sabry F. Girgis¹ (Department of Aeronautics, Faculty of Engineering, Al-Fateh University, Tripoli, Libya)

Rudder skin cracks were noticed on the lower part of most aircrafts of one series. The series of aircraft were grounded for rudder repair. Investigations were made later and a permanent solution to overcome this failure was devised. The stresses that cause failure depend on the dynamic characteristics of the panel and on the external excitation. The dynamic characteristics were determined by ground resonance tests, in the form of resonant frequencies and mode shapes. The external excitation was due to intense sound pressure generated by the jets of engines. The excitation of both turbulent boundary layer and vortex effects are at low level. Equivalent specimens of the rudder panel were constructed but with the different design approaches. Specimens were installed on elastic mountings and hinged to rigid frames in a similar manner to the lower part of aircraft

rudder of the jet trainer. The specimens were subjected to the same excitation of the jets of the engines. The duration of testings were accumulated to the same period in which the first crack was detected on the aircraft (100 h). Tests showed that improvement could be achieved by introducing intermediate support in the form of a spacer between the skins of the panel, or by increasing the number of stiffeners or ribs, or by just enlarging the rib flanges and adding one more rivet row. Initial tension of the skin also had a better effect, for example, applying air pressure between the skins produced such tension. Alternatively the reduction in skin stiffnesses caused by jet temperature had an adverse effect. Extensive theoretical calculations for panel instabilities under various support conditions were made. There was a close relationship between the static buckling mode and the mode causing acoustic fatigue failure. ¹Previously a research worker at Department of Research and Development, Helwan Aircraft Factory, Arab Organization for Industry, Egypt.

2:50

DDD4. Annoyance to advanced turboprop aircraft flyover noise. David A. McCurdy (NASA Langley Research Center, Mail Stop 463, Hampton, VA 23665)

A laboratory experiment was conducted to quantify the annoyance to advanced turboprop aircraft flyover noise. A computer synthesis system was used to generate 45 realistic, time-varying simulations of propeller aircraft flyover noise in which the harmonic content was systematically varied to represent the factorial combinations of five fundamental frequencies ranging from 67.5 to 292.5 Hz, three frequency envelope shapes representing helical tip Mach numbers of 0.63, 0.73, and 0.78, and three tone to broadband noise ratios of 0, 15, and 30 dB. In the experiment, 64 subjects judged the annoyance of recordings of the 45 synthesized flyover noises presented at D-weighted sound pressure levels of 70, 80, and 90 dB in a testing room which simulates the outdoor acoustic environment. Analyses of the judgments examine the effects on annoyance of the differences in the harmonic content of the flyover noises. The annoyance prediction ability of various noise measurement procedures and corrections is also examined.

3:05

DDD5. Alternative concept for aircraft interior noise control. Curtis I. Holmer (E-A-R Division, Cabot Corporation, 7911 Zionsville Road, Indianapolis, IN 46268)

Current technology for aircraft interior noise control depends on the use of weighted barriers and absorbing layers to control noise radiation into the cabin environment. This ignores the presence of structureborne flanking paths to interior structural/decorative surfaces which contribute significantly to the total cabin noise level. An alternative approach is to employ structural damping with (or without) vibration isolation to reduce noise radiated. This concept has significant potential for reducing treatment weight, because it can make use of the weight of decorative surfaces for acoustic functions. In this paper we review the results of a flight test demonstration program for this damped trim panel concept. The treated aircraft (a 12-14 passenger business jet) exhibited equal or lower noise levels to that achieved with weighted-layer treatments with more than a 40% reduction in the weight of acoustical materials used.

3:20

DDD6. Vibroacoustic habitability of space stations. David G. Stephens (NASA Langley Research Center, Mail Stop 463, Hampton, VA 23665)

The prediction and control of noise and vibration is essential in the design and operation of future space stations to ensure maximum crew efficiency and utilization. On a number of previous space vehicles, noise and vibration levels have exceeded design specifications and have evoked crew complaints, particularly in the areas of sleep and communication, thus suggesting the need for better vibroacoustic prediction and control techniques. In other cases, adverse reactions have resulted despite design specifications being met which suggests that earth based design criteria may be inadequate for space applications. A technology program is being formulated to develop prediction methods and control techniques for use in the design and operation of space stations to ensure acceptable vibroacoustic levels for hearing, communication, performance, comfort, and sleep. The goals include: accurate vibroacoustic prediction and assessment models for space station design; criteria for assessing the effects of different space station crew exposure levels; and new techniques for controlling the vibroacoustic environment within a space station.

3:35

DDD7. Qualification of a hemi-anechoic room for sound power level determination. Bart L. Burkewitz (E-A-R Division, Cabot Corporation, 7911 Zionsville Rd., Indianapolis, IN 46268)

As part of E-A-R's technical expansion program for its noise and vibration control laboratory, an interim hemi-anechoic room has been added to the facilities. Consistent with collection of quality acoustic data, special acoustic environments need to be understood. To accomplish this, ANSI Standard S1.35-1979, "Precision Methods for the Determination of Sound Power Levels of Noise Sources in Anechoic and Hemi-Anechoic Rooms," was selected as both a tool to guide study of the room and a documentation procedure to provide assurance to subsequent recipients of results of tests performed in the room. This paper will discuss the specific test procedures used to qualify the room under the standard, data analysis techniques used, and the idiosyncracies involved in performing tests of this type.

3:50

DDD8. Document printer source noise control—A case history. Daniel T. Lilley (E-A-R Division, Cabot Corporation, 7911 Zionsville Road, Indianapolis, IN 46268)

Noise reduction of at least 17-dB A-weighted sound level was obtained on a multi-function, dot matrix document printer by incorporation of damping, vibration isolation, and modification of the acoustical paths within the machine. This result was achieved without the use of sophisticated measurement and analysis techniques, but by consideration of basic principles of structural dynamics, application of appropriate materials in conjunction with minor design modifications, and attention to detail. Method of measurement was per ANSI S1.29-1979, "Method for the Measurement and Designation of Noise Emitted by Computer and Business Equipment." Emphasis is given to the importance of using highly damped isolation materials which permit large impedance mismatches at structural attachment sites while maintaining integrity of static connections. A discussion of noise sources and treatment approach for each is provided.

4:05

DDD9. Turbulent boundary layer beneath blankets: Measurements and interpretations. F. E. Geib and G. Maidanik (David Taylor Naval Ship Research and Development Center, Code 942, Bethesda, MD 20084)

An analytical formulation of the performance of a blanket as a spatial-temporal filter for turbulent boundary layer pressure fluctuations is pre-

sented. The formulation is developed as an aid for the design of experiments and experimental procedures for measuring the performance of blankets. The interpretation of data acquired in such experiments is treated in some detail. A simplified analytical model, which has the gross features of the pressure field beneath a convecting turbulent boundary layer, is utilized. This serves to illustrate the trends that can be anticipated in data and to illustrate the effects of undesirable stray pressure fields. Some limited data were acquired in a quiet turbulent pipe facility and are treated with the developed formalism. Significant features in the data are briefly discussed.

4:20

DDD10. Water-flow-induced tones associated with resonant fluctuations of a free flooding cavity. S. A. Elder (Physics Department, U. S. Naval Academy, Annapolis, MD 21402)

A towed-model apparatus designed to study nonresonant water-flow-induced cavity tones was found to exhibit strong fixed-frequency oscillation at certain speeds. Stationary ensoufflement of the model by means of a J9 projector disclosed six apparent resonances in the range 0–200 Hz, five of which could be excited by flow over the cavity. Accelerometer measurements of the cavity wall vibration are consistent with the hypothesis that the acoustic oscillation is due to fluctuations in the chamber cross section. The strongest mode was found to occur at a Strouhal number predictable by a formula previously derived for air pipetones. [Work supported by Naval Sea Systems Command General Hydromechanics Research Program, administered by the David W. Taylor Naval Ship R & D Center, Bethesda, MD.]

4:35

DDD11. Measured and predicted features of the quasisteady wall pressure field beneath turbulent spots in a laminar boundary layer. Fred C. De Metz (11815 Jellico Avenue, Granada Hills, CA 91344)

The measured features of the quasisteady components of the wall pressure field beneath naturally and artificially generated turbulent spots in a laminar boundary layer are compared with those calculated from structural models based on the large-scale motions within the spots. Contrasts reported in the literature concerning whether the wall pressure disturbance during spot passage is initially negative or positive and the spatial distribution of the spot's wall pressure field are discussed [Cantwell, Coles, and Dimotakis, *J. Fluid Mech.* **87**, 641–672 (1978); Mautner and Van Atta, *J. Fluid Mech.* **118**, 59–77 (1982)].

4:50

DDD12. Correlation of flight effects on centerline velocity decay for cold-flow acoustically excited jets. Uwe H. von Glahn (NASA Lewis Research Center, Cleveland, OH 44135)

Acoustic excitation can influence the large-scale shear layer structure of jet flow, thereby causing the jet centerline velocity to decay more rapidly. This phenomena has numerous practical applications, such as the reduction of jet/flap impingement velocity in order to reduce flap structural loads for under-the-wing STOL aircraft concepts. In the present paper, cold-flow centerline velocity decay data obtained with acoustic excitation of the jet exhaust flow are correlated with and without flight effects. Initially, static data are correlated in terms of a new acoustic parameter involving the threshold sound level needed to initiate acoustic excitation of the jet plume. It is shown that for a given flight speed, the same acoustic excitation parameter is valid as that developed for the static condition. Finally, an aerodynamic flight effects parameter is included to correlate the static and flight centerline velocity decay data with and without acoustic excitation.

Session EEE. Architectural Acoustics VIII and Musical Acoustics VIII: Architectural Acoustics for the Organ

Ewart A. Wetherill, Chairman

Bolt Beranek and Newman Inc., 21120 Vanowen Street, Canoga Park, California 91303

Chairman's Introduction—1:30

Invited Papers

1:35

EEE1. **Architectural acoustics for the organ.** Robert B. Newman (Bolt Beranek and Newman Inc., Cambridge, MA 02238)

Organs are used in many settings: churches, concert halls, recital halls, and even practice rooms. An organ may be used as a solo instrument, as accompaniment to a chorus, or as part of an orchestral ensemble. There is a wide range of appropriate acoustics for the auditorium as well as placement of the instrument and performer with relation to the audience and to the other performers. These matters will be discussed.

2:05

EEE2. **Preliminary observations of the influence of casework of pipe organs on room acoustical fields.** Bertram Y. Kinzey, Jr. (Department of Architecture, University of Florida, Gainesville, FL 32611)

A studio organ was built for the University of Florida so that its casework can enclose the pipework except for the open front or be removed to permit pipes to be free-standing. The sound of pipes in various locations was recorded at a number of points in the room with and without the case in place. The university organist recorded the same Bach prelude and fugue using identical registration with and without case enclosure. Data are presented showing the comparative spectral analysis of selected notes with and without casework. Professional musicians were asked to pick the cased and uncased examples of the recorded prelude and fugue and give their reactions. Generally, the sound with the case in place contained louder lower harmonics relative to higher pitches than sound from free-standing pipes. Musicians readily differentiated between the cased and uncased examples of the prelude and fugue and observed a better blending of encased high-frequency notes which was lacking for free-standing pipes.

2:20

EEE3. **Room acoustics for organs.** Bertram Y. Kinzey, Jr. (Department of Architecture, University of Florida, Gainesville, FL 32611)

Traditional criteria of very reverberant environments for organ music are cited. These are compared to such considerations as small instruments in small spaces vs larger ones in large volumes. The need for clarity in the audition of the polyphonic literature for the organ is discussed and compared to the requirement of speech intelligibility in churches where good acoustics for both music and speech are a goal. All the factors presented relative to acoustical environments for organs indicate the need for a broader range of criteria than organists and organ builders and others typically accept.

2:35

Presentations by Panel of Experts

Participants

Daniel W. Martin

Baldwin Piano and Organ Company, 1801 Gilbert Avenue, Cincinnati, Ohio 45202

Ronald F. Ellis

M. P. Moller, Inc., 403 N. Prospect Street, Hagerstown, Maryland 21740

Jean-Louis Coignet
*Organier-expert for the City of Paris, France, Casavant Freres Ltd., C. P. 38, St.
Hyacinthe, Quebec, Canada J2S 7B2*

Manuel J. Rosales
Rosales Organ Builders, Inc., 160 Glendale Blvd., Los Angeles, California 90026

Duane A. Kuhn
*Classical Organ Division, Baldwin Piano and Organ Company, 1801 Gilbert Ave-
nue, Cincinnati, Ohio 45202*

Lawrence Phelps
Allen Organ Company, Macungie, Pennsylvania 18062

Contributed Paper

3:35

EEE4. Electroacoustic assistance of parish church organ music. R. L. Prater, F. W. Christie, and C. Stockbridge (American Bell Inc., E. D. & D., Holmdel, NJ 07733)

When the new organist at St. George's-by-the-River church in Rumson, New Jersey complained that his greatest efforts at filling the church with music seemed futile because the reverberation time was too low, we attempted to quantify this feeling and explore low-cost remedies. The church is of modest size, a 8.1×9.6-m chancel area and a 19.7×17.8-m

nave. We measured an average reverberation time near 1.5 s (measured in third octave bands up to 8 kHz), observed inefficient coupling of sound energy from the pipe organ chamber via the chancel to the congregation, due to location of the sole acoustic feed high in the side wall of the chancel. To augment the acoustic coupling a microphone was positioned in front of the organ chamber, and the signal was digitally delayed by 82 ms, amplified, and fed into a pair of powerful loudspeakers positioned high and against the rear corners of the nave. This produced a substantial increase in sound level (> 6 dB over 3 octaves). Both we and the organist found the natural character of the pipe organ with this mild electroacoustic assist quite gratifying.

3:50 to 4:45

Panel Discussion

FRIDAY AFTERNOON, 11 NOVEMBER 1983

GARDEN ROOM, 2:00 TO 5:07 P. M.

Session FFF. Speech Communication VIII: Speech Physiology and Timing

Peter J. Benson, Chairman
ITT-DCD, 10060 Carroll Canyon, San Diego, California 92131

Chairman's Introduction—2:00

Contributed Papers

2:05

FFF1. Electroglottographic and acoustic waveforms of voice onset in stutterers. Gloria J. Borden and Thomas Baer (Haskins Laboratories, New Haven, CT 06510)

The first few glottal pulses in the stuttered and fluent tokens of voiced utterances by stutterers were analyzed and compared with the initial voice pulses in the same utterances by nonstutterers. Signals analyzed were the impedance changes across the glottis during voice initiation via electroglottography (EGG) and the acoustic waveforms. In single-cycle analysis of vocal fold vibration, normal speakers evidenced abrupt decrease in impedance with increasing vocal fold contact, and a more gradual increase in impedance as the folds opened. Stutterers, when perceived as fluent, followed the same pattern as normals. Extremely aberrant EGG

patterns often accompanied stuttering episodes, with idiosyncratic strategies revealed for breaking the stuttering block. The envelope for the first EGG cycles differed from normal for some stutterers. Some of the severe stutterers showed patterns of gradual buildup of EGG amplitude after a block, whereas the mild stutterers and normal controls showed a more abrupt envelope. When fluent, all speakers had EGGs with abrupt envelopes and a more stable open phase for each cycle. EGG activities will be related to their acoustic correlates. [Work supported by NIH.]

2:17

FFF2. An aerodynamic study of stops with different phonation types. Sarah N. Dart and Patricia A. Keating (Phonetics Lab, Department of Linguistics, UCLA, Los Angeles, CA 90024)

Measurements were made of intraoral air pressure and oral airflow during the production of stops with different phonation types. The sounds investigated include Korean lenis and fortis, and Hindi voiced and murmured, stops. An aerodynamic model was used to simulate possible articulations which could produce air flows and pressures like those observed. For example, higher intraoral pressure and lower oral flow were found for Korean fortis as compared to lenis stops. This difference follows from a simulated difference in vocal tract wall tenseness. Furthermore, the right shape of intraoral pressure curves can be simulated with a difference in subglottal pressures. This analysis-by-synthesis approach, combining physiological measurements and articulatory modeling, provides new insights into differences in phonation types. [Work supported by USPHS grant NS 18163-02 to Peter Ladefoged.]

2:29

FFF3. Control of rate and movement duration in speech. David J. Ostry, Kevin G. Munhall, and Avraham Parush (Department of Psychology, McGill University, Montreal, Quebec H3A 1B1, Canada)

This paper provides evidence on the control of rate and movement duration in speech. A computerized pulsed-ultrasound system was used to monitor separately tongue dorsum movements and laryngeal gestures during the production of CV and CVCVC sequences. The kinematics of tongue and laryngeal movements were analysed by partitioning the lowering gesture of the tongue and both the abduction and adduction gestures of the vocal folds to give estimates of displacement, duration, and maximum velocity. For both articulators the ratio of the maximum velocity to the extent of the gesture was found to vary inversely with the duration of the movement. The finding suggests that a single function might account for a wide range of changes in the duration of individual gestures. The control of movement rate and duration through the regulation of biomechanical characteristics of speech articulators is discussed.

2:41

FFF4. Ultrasonic orientation methods for tongue scanning. Sandra L. Hamlet (Department of Hearing and Speech Sciences, University of Maryland, College Park, MD 20742)

Real-time ultrasonic imaging of the tongue presents a problem in verification of the orientation of the scan, especially when detailed measurements are to be made, or replication of measurements is attempted. Ultrasonically identifiable landmarks may be free to move (e.g., hyoid bone) and thus present an inherently unstable reference. A review and evaluation of various orientation techniques is presented, including (1) use of external facial features for transducer placement, (2) use of ultrasonically distinguishable anatomy, (3) placement of oral markers or probes, (4) adoption of standard speech and speech-like gestures. [Work supported by NIH.]

2:53

FFF5. Jaw muscle activity for speech and nonspeech gestures. Michèle Gentil and Thomas Gay (Department of Oral Biology, University of Connecticut Health Center, Farmington, CT 06032)

The activity patterns of the mandibular elevator muscle system were investigated for speech and nonspeech gestures for the purpose of determining the possibility of neuromuscular specialization for speech in relation to other mandibular functions. Intra-muscular wire electrodes were placed into the seven muscles of the mandibular system and the anterior belly of the digastric. The activity of these muscles was recorded along with a time varying magnetometer voltage that corresponded to the displacement of the jaw in two and three dimensional space. These signals were recorded from three adult American English subjects during the production of speech at different rates and for nonspeech gestures that included border movements, functional opening-and-closing and mastication. Results indicated that jaw space for speech is the most restricted of any mandibular function, jaw movements for speech are generally produced by the most simple muscle action patterns and individual differences exist in the selection of a given muscle or muscle system for a specific

function and/or utterance, and in the coordination of muscle activity patterns for any particular gesture. [Research supported by a grant from the National Institute of Health (NS-10424).]

3:05

FFF6. Vibratory patterns of the vocal folds during pulse register phonation. Robert L. Whitehead, Dale E. Metz, and Brenda H. Whitehead (National Technical Institute for the Deaf, P. O. Box 9887, Rochester, NY 14623)

In a previous paper [R. L. Whitehead *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 73, S47 (1983)] we reported that during a single vowel sample of pulse register phonation, each vibratory cycle of the vocal folds consisted of either a double or triple pulsing motion prior to achieving complete glottal closure. The current paper presents data on two additional vowel samples obtained on the same female subject phonating in pulse register. Glottal area-time functions were calculated, using high speed laryngeal films, for 35 consecutive cycles from one film and for 33 consecutive cycles from the other. The results from the first film indicated that each vibratory cycle of the vocal folds consisted of a single opening/closing gesture followed by a lengthy closed period. The results from the second film indicated that each vibratory cycle consisted of a double opening/closing vocal fold pattern followed by the lengthy period of closure. From our data, it appears that one of the physiological descriptors of pulse phonation is that multiple, as well as single, vibratory patterns of the vocal folds. [Work supported by U. S. Department of Education.]

3:17

FFF7. Stability analysis of the human vocal cords. Charles Thompson, J. Wallace Grant, and Sharon A. Starowicz (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

A mathematical model will be presented to describe the behavior of the vocal cords during the production of speech. The model for the folds consists of two-hinged gates which serve to modulate the flow through the glottis. The degree of flow modulation is related to the angular displacement of the gates. Equations governing the mechanical motion of the gates are derived by satisfying conservation of mass and momentum across the glottis. The introduction of acoustic impedance of the vocal tract serve to couple mechanical, acoustic, and fluid dynamic elements of the model. A linear stability analysis will be presented detailing the range of mechanical constants for which the motion of the gate will become stable. It is shown that stability of the gate motion is not dependent on the trans-glottal pressure distribution but on the acoustic-mechanical interaction.

3:29

FFF8. Differential negative resistance in a one-mass model of the larynx with supraglottal resistance. William A. Conrad (30 W. 71 Street, New York, NY 10023)

The steady flow calculations of Conrad [Conference on Physiology and Biophysics of Voice, Iowa (1983)], show that the larynx can manifest a differential negative resistance, i.e., translaryngeal pressure can decrease as flow rate increases. These calculations were based on the one-mass model of Flanagan and Landgraf [IEEE Trans. Audio Electroacoust. AU-16, 57-64 (1968)] with added supraglottal resistance. If the spring acting on the mass is changed from linear to nonlinear, the differential negative resistance becomes bounded by two regions of positive differential resistance. As nonlinear stiffness increases, differential negative resistance disappears. The one-mass model plus supraglottal resistance can be described as an *N*-type, flow-controlled nonlinear resistance, QNLR. It is well known in electronics and elsewhere that a QNLR in a series resonant circuit can transform dc power to oscillatory power [e.g., Ph. le Corbeiller, Proc. Inst. Elec. Eng. 79, 361-378 (1936)]. The system can be usefully analyzed in terms of van der Pol's equation which gives the oscillation condition, the frequency, rate of build-up, limiting amplitude and efficiency. The analysis shows that differential negative resistance is a necessary condition for oscillation.

Contributed Papers

3:55

FFF9. Durational characteristics of vocal and subvocal speech. Bruce L. Smith and James Hillenbrand (Speech and Language Pathology, Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

In an attempt to investigate issues related to speech timing, adult subjects were asked to perform two tasks. They produced a number of disyllabic words and phrases orally, and they also "thought" the same words and phrases without vocalizing them. Some of the stimuli were easy to pronounce (e.g., "baseball"), while others were more difficult (e.g., "wrist watch"). Subjects repeated each stimulus 20 times in succession as rapidly and as accurately as possible. Durations of vocal and subvocal productions were measured by having subjects tap the space bar of a computer terminal each time they said or thought the word. Measurements of inter-tap intervals suggest that stimuli which were more difficult to pronounce vocally (indicated by longer inter-tap intervals) were also more difficult to produce subvocally ($r = 0.86$). However, subvocal durations averaged about 15% shorter than oral productions. These findings appear to support models of speech production which propose that speech segments have temporal characteristics associated with them at the abstract organizational level, but the results also seem to show output-level effects as well.

4:07

FFF10. Acoustic characteristics of flap productions by American English-speaking children and adults. Robert Rimac and Bruce L. Smith (Speech and Language Pathology, Northwestern University, 2299 Sheridan Road, Evanston, IL 60201)

Acoustic characteristics of underlying, intervocalic /d/ and /t/ flaps produced by a number of children and adults were studied to determine whether children's flaps showed similar properties to their other speech segments when compared with adults' productions. Closure durations of flaps and other consonants and a number of vowel durations were measured using a laboratory mini-computer system. Contrasts such as "pudding/putting" and "writer/rider" produced in a carrier phrase were elicited using pictures so that no auditory model would be available for the children. Preliminary results showed that flaps from underlying /d/ were about 10% longer in duration than flaps from underlying /t/ for both the children and adults. Although stressed vowels preceding underlying /d/ flaps were longer in duration than those preceding underlying /t/ flaps for both groups, this contrast was substantially greater for the adults. It was also observed that flap durations were approximately 40% longer for the children than for the adults, while the durations of the children's vowels and nonflap consonant productions were more comparable to the adult's durations for those segments.

4:19

FFF11. Effects of phrasing and word emphasis on transitional movements—Location and stability of tongue blade iceberg patterns. O. Fujimura and W. R. Spencer (Bell Laboratories, Murray Hill, NJ 07974)

In previous ASA meetings, it was suggested that certain movements of an articulator that is crucial for both CV and VC identities were often observed unaffected in their nearpeak parts as stable fragmental time functions, even when large variation in other portions such as vowel portions was introduced by prosodic control. By an automatic procedure, we have identified the parts of demissyllabic transitions that are likely to be stable under manipulation of phrasing such as $(21 + 2) \times 3$ vs $21 + (2 \times 3)$, and contrastive emphases placed on different words in a sentence such as "It's 9 8 11 Brooks Drive." The data were obtained by the computer controlled x-ray microbeam system at the University of Tokyo [Kiritani, Itoh, and Fujimura, *Acoust. Soc. Am.* 57, 1516-1520 (1975)], and pertain, in this paper, to the vertical position of a metal pellet placed on the tongue blade.

4:31

FFF12. Computation of mapping from muscular contraction patterns to formant patterns in vowel space. Y. Kakita (Kurume University School of Medicine, Kurume, Japan) and O. Fujimura (Bell Laboratories, Murray Hill, NJ 07974)

In previous papers [*J. Acoust. Soc. Am. Suppl.* 1 62, S15 (1977); in *Frontiers of Speech Communication Research*, edited by B. Lindblom and S. Ohman (Academic, London, 1979)], pp. 17-24, we demonstrated that a three-dimensional static model of the tongue can be used to explain some basic characteristics, in particular an insensitivity of the formant pattern of the vowel [i] to changes in the magnitude of muscle contraction force over a wide range when an appropriate set of muscles is used. We have implemented a computer program to compute systematically the mapping relation between various combinations of muscle contractions and the resultant formant frequencies in the vowel space. Muscular modeling for different vowels are being experimented based on available anatomical and physiological data, in particular those reported by K. Honda *et al.* [*J. Acoust. Soc. Am. Suppl.* 1 72, S103 (1982)]. Eventually, after the work by M. V. Mathews *et al.* [*J. Acoust. Soc. Am.* 63, 1535 (1978)] on mapping between area functions and formant patterns, this would illustrate inherent characteristics of the inverse relations from *F* patterns to muscular patterns for vowels. Uniqueness problems will be discussed with quantitative examples.

4:43

FFF13. Components of lingual and labial articulation. Jan Edwards, Katherine S. Harris (Graduate Center, CUNY and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510), and Betty Tuller (Cornell University Medical College, Ithaca, NY 14853 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

One problem in speech production research is that it is difficult to relate muscle activity in individual articulators to the resulting movements. In large part, this is because movement of any given articulator is highly dependent on movement of connected structures. Both tongue blade and lip movements contain a component due to jaw movement in addition to a component that reflects muscle activity of the tongue or lip. Using data obtained by the x-ray microbeam system [Kiritani *et al.*, *J. Acoust. Soc. Am.*, (1975)], tongue blade and lip movements were separated into these two components. Dynamic characteristics of tongue blade articulation of /i/ and /a/ and labial articulation of /p/ were examined across multiple tokens of two to four syllable nonsense utterances. The timing of lip and tongue blade articulations relative to jaw raising for /p/ and jaw lowering for /i/ and /a/ is influenced by surrounding phonetic context and syllable stress. It was found that the lip and the tongue blade compensate for some but not all of the variations in inter-articulator timing and positioning. [Work supported by NINCDS.]

4:55

FFF14. The word-position similarity constraint on segmental speech errors: Further experimental results. Stefanie Shattuck-Hufnagel (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

In earlier experiments it was shown that speech errors involving the interaction of two target segments are governed by a word-position similarity constraint: two segments are more likely to interact in an error if they are word initial then if they are stressed-syllable initial or simply syllable initial. Further experiments have shown that this effect (1) is more pronounced for nonwords than for words, suggesting that the phenomenon arises during the computation of a short-term representation of the units of the utterance rather than during lexical access, and (2) is found in sentence generation tasks as well as in the original tongue-twister repetition tasks, making an explanation in terms of memory constraints less likely and an account in terms of production planning more likely. [Work supported by NIH-NINCDS Grant #2 RO 1 NSO4332 19.]

Session GGG. Underwater Acoustics VII: Signal Processing (Précis-Poster Session)

Harry A. DeFerrari, Chairman

Rosensteil School of Marine and Atmospheric Science, University of Miami, 4600 Rickenbacker Causeway, Miami, Florida 33156

Chairman's Introduction—1:00

Invited Papers

1:05

GGG1. High-resolution beamforming with oversampled arrays. Charles L. Byrne (The Catholic University of America, Washington, DC 20064) and Raymond M. Fitzgerald (Naval Research Laboratory, Code 5120, Washington, DC 20375)

Plane-wave beamforming is interpretable as estimation of the Fourier wave vector spectrum from samples of the spatially dependent acoustic field. Because spatially localized sources contribute discrete peaks to the wave vector spectrum, the use of high-resolution nonlinear techniques (such as Burg's maximum entropy method—MEM) to resolve closely spaced sources has received much attention [Johnson, Proc. IEEE 70(9), 1018–1028 (1982)]. Such techniques are expected to be most useful for short arrays, whose dimensions do not exceed a few wavelengths, for which case conventional beamforming offers poor resolution. Such short arrays typically contain a number of hydrophones and thus oversample the spatial field; the spacing between adjacent hydrophones is less than one half a wavelength. When MEM is applied to oversampled arrays, spurious features are often produced due to the concentration of the spectrum within the visible region of wave vectors. These spurious features can be eliminated through the use of a more general high resolution procedure, the weighted reciprocal spectrum approximation method [Byrne and Fitzgerald, IEEE Trans. Acoust. Speech Signal Process. ASSP-31, 276–279 (1983)].

1:30

GGG2. Signal design considerations for long-range shipboard active sonar. Thaddeus G. Bell (Naval Underwater Systems Center, New London Laboratory, New London, CT 06320)

If a long-range shipboard active sonar is to successfully cover target ranges out to the first convergence zone at 35 miles, an optimum frequency about 3 kHz is implied along with a usable band approximating 1.5 kHz. This modest band has many competing demands on its utilization besides signal design, some of which are subband allocations for: multiple-ship mutual interference, increased data rate, and adjacent sector coverage without reverberation buildup. The basic band segment available for signal design must be used to cope with either high or low Doppler echoes in a combination of noise and reverberation interference. Echoes differ significantly from the transmitted signal from spreading effects both in time and frequency. Time spreading can be produced by multiple paths and target structure. Frequency spreading can be produced by target and own ship motion. Reverberation spectra can be significantly influenced by wave motion, own ship motion, relative bearing of transmission and biological scattering resonances. Selected signal design options are considered for coping with the foregoing signal and interference characteristics. Methods of estimating consequent detection performance are presented.

Contributed Papers

1:55

GGG3. Source depth classification by modal decomposition and correlation. DeWayne White and Stanley Chin-Bing (Naval Research and Development Activity, Numerical Modeling Division, NSTL Station, MS 39529)

Various methods have been and are being used to classify and localize submarine targets using real-time information from acoustic sensors. We have investigated another technique which utilizes the environment and its variability to exclude (theoretically) all possible target depths except the correct one. The method involves synthesizing the normal modes using the best available sound speed profile and geoacoustic model as inputs to a complex normal mode model. Both single frequency and broadband fields can be modeled. These synthesized modes are sampled in depth (or range) to form a suite of replicas. The synthesized replicas are then correlated with the modally deconvolved incoming data having the same spatial sampling as the replicas. The ambiguity surface shows relatively high

correlation for replicas having the target's true depth. At present, synthesized data containing noise has been used for comparisons. However, the method shows promise of being a useful tool for depth localization. Examples will be presented for both vertical and horizontal arrays.

1:59

GGG4. A comparison of spatial spectra from vertical line arrays of different aperture using maximum entropy method (MEM) beamforming. Mimi Zebrick Lawrence (ODSI Defense Systems, Inc., 6110 Executive Boulevard, Rockville, MD 20852) and Dan J. Ramsdale (Naval Ocean Research and Development Activity, Ocean Acoustics Division, Code 345, NSTL Station, MS 39529)

This study was undertaken to provide a preliminary answer to the question of how aperture size affects beamwidth and resolution for vertical line array data which is processed using the (MEM) beamformer. This

was done empirically by examining the (MEM) beamformed output of a nine-element vertical line array and reducing the aperture by deleting hydrophones. Narrow-band acoustic data were provided by a cw source (54 Hz) which was towed at a nominal depth of 91 m radially away from the array. The (MEM) beamformer was implemented using the Burg algorithm in complex form with averaging performed over the filter coefficients. The spatial spectrum was characterized in general by two dominant arrivals which varied in angular separation and level relative to the background noise. For arrivals with high signal-to-noise ratios and large angular separation, preliminary results showed little degradation in beamwidth as the aperture was decreased by a factor of two. As the angular separation between arrivals decreased or when the signal-to-noise ratio decreased, beamwidth and resolution were degraded when the aperture was reduced.

2:03

GGG5. Response of a towed array to acoustic fields in shallow water. M. J. Buckingham¹ (Naval Research Laboratory, Code 5120, Washington, DC 20375)

An analysis of the response of a towed array to the modal field in isovelocity shallow water is presented. In general, the response consists of a *coherent* sum over all the modes. However, when the source is endfire-on to the array, or in that vicinity, the towing motion introduces a certain amount of range averaging into the response to the farfield radiation. A criterion is given, in terms of the interference length between the two lowest order modes in the field, for the degree of range averaging required to reduce the output of the array to an *incoherent* sum over the modes. This sum is associated with a set of virtual sources distributed in azimuth, each of which is associated with a particular normal mode. This means that some signal energy is rejected by the array, which results in a reduction in the signal gain. At the same time beam broadening, or in extreme cases beam splitting, arising from the angular spread of the virtual sources, occurs and this may give rise to a significant bearing error. ¹Currently on exchange from the Royal Aircraft Establishment, Farnborough, Hants, UK.

2:07

GGG6. A look at detection performance for maximum entropy beamforming. Arthur J. Collier (Code 5122, Naval Research Laboratory, Washington, DC 20375)

A statistical investigation of the detection performance of maximum entropy beamforming for low level signals is described. The investigation was performed for a horizontal line array with a half-wavelength frequency of 500 Hz. Both sparse and filled array configurations were investigated. Realistic models of a deep-water ambient noise field and array self-noise were used to define the expected cross-spectral matrix for the array in a noise field. Signals of 60, 200, and 400 Hz coming from near broadside and near end-fire were considered. The detection statistics were obtained by generating sample mean cross-spectral matrices with a Wishart distribution, using the Bartlett decomposition as implemented by Smith and Hocking [Applied Statistics, 1972]. The results are summarized in ROC curves.

2:11

GGG7. Digitizing effects on noise dynamic range and signal spectra in broadband arrays. M. D. Green, J. W. Young, and F. J. M. Sullivan (Bolt Beranek and Newman Inc., 3065 Rosecrans Place, Suite 208, San Diego, CA 92110)

Digitizing of the signals from elements in an acoustic array results in various types of errors in the beamforming process. Among these are quantization noise, clipping noise, spectral distortion, and false-target generation associated with clipping. This paper presents an analysis of these effects in the context of design requirements for a broadband receiving array. The array must operate over a specified dynamic range of ambient noise. A method is developed for ensuring that the sum of quantization noise and clipping noise will be kept at a specified level below ambient noise over the entire operating range. The optimum selection of quantiza-

tion levels for this purpose is found. When a signal exceeds the saturation level of the A/D converter, spurious signals will be generated at new frequencies, which may alias back upon the band of interest. When two or more signals jointly exceed the saturation level, intermodulation effects among signals having different arrival angles may produce spurious signals which are coherent across the array. These signals appear in frequency-wavenumber space as false targets. An analysis and a computer simulation of this effect are presented. The requirements on the minimum acceptable saturation level of the A/D converter are determined.

2:15

GGG8. Interpolation noise in a hard-limited beamformer. J. W. Young (Bolt Beranek and Newman Inc., 3065 Rosecrans Place, Suite 208 San Diego, CA 92110)

Beamforming typically requires sampling of individual hydrophone outputs at a rate substantially higher than Nyquist for bandpass processes. One approach to minimizing the sampling rate is the use of an interpolation filter for reconstruction of intermediate sample values [R. G. Pridham and R. A. Mucci, J. Acoust. Soc. Am. 63, 425-434 (1978)]. In principle, the required samples can be determined without error from the Nyquist rate samples since the latter contain complete information on the band-limited element signal. When the element signals are hard-limited, all amplitude information is lost and errors are introduced into the interpolation process. Information in a hard-limited signal (HLS) is contained in the zero-crossing times. Basically, sampling a HLS obliterates the information necessary to determine exact zero-crossing times between sample values. In this paper, we present an analysis of the mean square error (interpolation noise) associated with estimating the values of the HLS at intermediate times. We compute the increase in beam noise power due to this error and show that this imposes a limit on our ability to perform adaptive null steering with such an array.

2:19

GGG9. On the use of focused horizontal arrays as mode separation and source location devices in ocean acoustics-numerical modeling results James F. Lynch (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543) and Daniel K. Schwartz (Department of Physics, Harvard University, Cambridge, MA 02138)

It has been shown theoretically that one can perform normal mode separation and source location using focused horizontal arrays [J. F. Lynch, J. Acoust. Soc. Am. (to be published)]. To illustrate this theory, numerical modeling results are displayed for both line and convex circularly curved arrays in range-dependent and range-independent waveguide environments, both in the presence and absence of source motion. Various topics to be discussed include: direct measurement of modal group velocities from moving cw source measurements, determination of environmental range variations, mode interference effects and ray theory multipath focusing, and the intercomparison of focused array mode filtration with other standard techniques, e.g., steering.

2:23

GGG10. Initial sea test results of a multi-hydrophone digital buoy system with *in-situ* FFT processing capabilities. Peter C. King and Peter D. Herstein (Naval Underwater Systems Center, New London, CT 06320)

Two major problems have been historically associated with self-contained analog acoustic data acquisition buoy systems. First, analog tape recorder characteristics bound the obtainable dynamic range of the received data. Second, post-processing of the data tapes to obtain the required acoustic information can be both time consuming and expensive. To address these problems, a new buoy has been developed at NUSC. Acoustic data from each of eight hydrophones is digitized and then fast Fourier transformed by the buoy's micro-processor. Complex and ensemble averaged spectral results are then written onto a digital cassette tape. An instantaneous dynamic range of more than 72 dB is available over the frequency band 2-350 Hz. The spectral results are computed in 2 Hz

increments and Hanning weighting is used. The first at-sea test of the system occurred in August 1983 at a deep water location near Bermuda. In this paper we will present results from that test. Received signals from stationary and moving cw sources will be emphasized. [This work was sponsored by the Surveillance Environmental-Acoustic Support Project (SEAS), Naval Ocean Research and Development Activity, Code 520 (C. E. Stuart, Program Manager).]

2:27

GGG11. Sidelobe level of hyperbolic frequency modulated (HFM) waveforms. S. I. Chou (Naval Ocean Systems Center, San Diego, CA 92152)

As large time-bandwidth products are employed for the widely used LFM (linear frequency-modulated) waveforms, the loss due to the phase mismatch of the peak output of a single matched filter becomes excessive even for very moderate target radial speeds. HFM (hyperbolic frequency modulated) waveforms can eliminate this loss. An early result on sidelobe level of HFM waveform [J. J. Kroszczynski, Proc. IEEE 57, 1260-1266 (1969)] apparently has errors in its derivation in applying asymptotic approximation techniques. The only numerical result provided there did not agree well with the analytically derived result, despite the claim to the contrary. In the present paper, new asymptotic results have been derived using integration by parts in the correct direction, so that terms discarded are smaller than the quantity being approximated. Computer simulations using several sets of parameters, including the one mentioned above, compare well with the new analytic approximations. The increased sidelobe level of HFM waveform over that of the LFM one is much less than indicated by the early approximation.

2:31

GGG12. Array gain improvements at low SNR and low frequency via maximum entropy beamforming. Donald R. Del Balzo (Naval Research Laboratory, Code 5120, Washington, DC 20375)

It is fairly well accepted that nonlinear beamformers will significantly improve array performance for the case of moderate to high input SNR and near the design frequency. Most applications of high-resolution beamformers have been target resolution and localization, assuming an initial detection has been made. This work addresses array gain at low SNR for blade rate frequencies well below design using a maximum entropy beamformer (MEB) on a horizontal, bottomed array. Simulation results show array gain improvements for the MEB, when compared to a conventional approach, that depend on the input SNR and averaging time.

2:35

GGG13. A ray theory simulator of a moving broadband source. James D. Ratliff and Clark S. Penrod (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

A computer model is described which simulates and processes the time series received from a moving broadband source. Ray theory is used to compute the set of eigenrays instantaneously valid at each source emission time in a range invariant environment characterized by an arbitrarily depth-dependent water-column sound-speed profile atop a single sediment layer with linear sound speed and attenuation profiles overlying a semi-infinite homogeneous substrate. The received time series is obtained by a discrete time domain convolution of an arbitrary source waveform with a set of impulse responses which depend on emission time and propagation path. The received time series can be autocorrelated or crosscorrelated to examine the impact on broadband signal processing due to environmental (particularly bottom interaction) effects, source motion effects (Doppler and multipath structure evolution), and the choice of signal processing parameters. Example correlation functions are shown to illustrate the effects of some of these mechanisms. [Work supported by Naval Ocean Research and Development Activity, Code 530.]

2:39

GGG14. The influence of environment and source motion on broadband acoustic correlations: A normal mode approach. Robert Gragg (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029)

We deal with a source which emits a broadband signal $S(t)$ while moving through a shallow (<300 m), range-invariant environment at constant speed, depth, and bearing. The received signal is constructed from the result [K. E. Hawker, J. Acoust. Soc. Am. 63, 675-681 (1979)] for the $S(t) \sim e^{i\omega t}$ case. In terms of Fourier transforms,

$$P(r, z, t) \sim \sum_n \hat{S}(\omega) \exp\{i[k_n(\omega) - \delta_n(\omega)]\} \\ \times [1 - \epsilon_n(\omega) \cos \theta(t)] |r - r_0(t)| \\ \times U_n^*[z, \omega_n(\omega, t)] U_n[z_0, \omega_n(\omega, t)]'$$

is the signal at r, z, t from a source on trajectory $r_0(t)$ at depth z_0 and speech $\theta(t)$ is the source-receiver angle, $\epsilon_n(\omega) = v_0/v_n^p(\omega) \ll 1$, and $\omega_n(\omega, t) = \omega - v_0 k_n(\omega) [\cos \theta(t) - \epsilon_n(\omega)]$ is a Doppler-shifted frequency. We model subkilohertz-band signals in this way for fixed explosive sources (SUS) and for moving noise sources. The signal and its autocorrelation are compared with experimental data to determine the importance of environmental factors (gross SVP features, finer SVP details, bottom attenuation profile) and source motion (speed and CPA range), as well as source and receiver depths. [Work supported by DARPA TTO and Naval Electronics Systems Command (Code 612).]

2:43

GGG15. Cross correlation of two hyperbolic-frequency-modulated (HFM) waveforms. S. I. Chou (Naval Ocean Systems Center, San Diego, CA 92152)

In a multi-user radar/sonar environment, or in the case of a single user but with different pulses transmitted, it is important to know the cross correlation of the waveforms as a function of time delay. The cross-correlation property of two differently swept LFM (linear-frequency-modulated) waveforms is straightforward from the view point of using the stationary-phase approximation. There, as one waveform's frequency-time curve slides against the other's, the intersecting angle remains constant over an interaction interval in time delay corresponding to the common frequency interval of the two waveforms. The picture changes vastly when the two LFM waveforms are replaced by two HFM (hyperbolic-frequency-modulated) ones. In this paper, the cross correlation over such an interval is established. Using examples, the analytical approximation is shown to compare favorably with numerical simulation results. The difference between the analytic and numerical results is further characterized using the unity volume constraint for narrow-band cross-ambiguity function which is lesser known than the auto-ambiguity counterpart.

2:47

GGG16. Coherent wavefront resolution. Donald A. Murphy (1062 Valencia Mesa Drive, Fullerton, CA 92633)

Resolving coherent arrivals is fundamentally different from resolving arrivals from independent sources because an interference field is created by the coherent arrivals. This results in a field which is not spatially stationary. At high signal-to-noise ratios considerably better resolution is possible than for the case of independent sources. An array which has a null of the interference field at its center will resolve the two arrivals for apertures considerably smaller than expected by using the Rayleigh criterion. The phase difference of the two arrivals is critical to resolution in this case. Resolution of this type was observed with signals which were received at a fixed array in the deep ocean from a single low-frequency source over long propagation paths through complicated sound speed structures, thus demonstrating the coherence of the multipaths involved. The coherence of the vertical multipaths follows from the discrete nature of the arrival structure. In this case each multipath is not appreciably spread in arrival angle.

GGG17. Sensitivity of a passive horizontal-tracking algorithm to input-measurement errors. P. Bilazarian,¹⁾ W. L. Siegmann, and M. J. Jacobson (Rensselaer Polytechnic Institute, Troy, NY 12181)

We consider the passive determination of bearing, range, depth, frequency, and horizontal velocity for a narrow-band acoustic source which moves at constant depth. An algorithm for tracking a moving source is developed and employed, using a horizontal linear array which is towed at fixed depth. The source and receiver move with constant horizontal velocity in the upper portion of a deep ocean and are separated by relatively short range. Acoustic signals are presumed to arrive along two upper-ocean ray paths. The tracking is accomplished using a new combination of processed multipath information from the ray paths, frequency shifts from a cw emission, and directional measurements from the linear array. The sensitivity of predictions for three-dimensional position, frequency, and horizontal velocity are demonstrated for small variations in environmental parameters and acoustic measurements. Results for a variety of source-receiver configurations are discussed. Variance estimates of position and motion predictions are obtained in terms of input measurement variances. [Work supported by ONR.]¹⁾ Present address: Siena College, Loudonville, NY 12211.

GGG18. Eigenvalue detection threshold statistics. Peter C. Mignerey (Naval Research Laboratory, Code 5120 Applied Ocean Acoustics, Washington, DC 20375)

A promising method for high-resolution beamforming on underwater arrays is the eigenvector decomposition method which requires knowledge of the expected noise in order for multiple eigenvalues to occur. [D. H. Johnson and S. P. DeGraaf, IEEE Trans. Acoustics Speech Signal Process. 30, 638 (1982).] It is presently not well understood how to objectively separate the signal and noise subspaces. When the correlation matrices are estimated with a finite number of degrees of freedom the rank becomes full and the eigenvalues spread. Under these conditions the classification of eigenvalues may be made using statistical methods. A Gaussian process was applied to the hydrophones of a ten-element filled array and sample noise matrices were generated in the metric of other independent sample noise matrices. Probability densities were obtained based on 10^6 samples for each eigenvalue and the broadside beam of conventional, maximum likelihood and maximum entropy beamformers. A comparison of ROC curves demonstrates sufficient detection capability by the eigenvalue method to allow the establishment of detection thresholds for the objective classification of eigenvalues.

FRIDAY AFTERNOON, 11 NOVEMBER 1983 SENATE/COMMITTEE ROOMS, 1:00 TO 4:35 P.M.

Session HHH. Underwater Acoustics VIII: Scattering and Noise

Herman Medwin, Chairman

Department of Physics, Naval Postgraduate School, Monterey, California 93940

Chairman's Introduction—1:00

Contributed Papers

1:05

HHH1. Surface statistics and the spatial coherence of surface-scattered energy. W. A. Kinney (NORDA Code 340, NSTL Station, MS 39529) and C. S. Clay (Department of Geology and Geophysics, Weeks Hall, University of Wisconsin, Madison, WI 53706)

The facet-ensemble method [W. A. Kinney *et al.*, J. Acoust. Soc. Am. 73, 183-194 (1983)] was used to determine how the statistics of a rough surface relate to the spatial coherence of energy scattered from it. Two ideal water surfaces were studied. One had a sawtooth cross section derived from the sum of the first four terms of a Fourier sine series representation. The cross section of the other surface was derived from the sum of the same terms but with each having random phase. This resulted in two surfaces each with the same rms roughness (0.2 cm), spatial wavelength (16 cm), and spatial wavenumber components, but different cross-sectional shapes. With each surface, the acoustic field in water was computed at two receiver locations for both backscatter and forward scatter for a grazing angle of 35 (with respect to the surface mean), 100 kHz, and a total travel distance of 421 cm. For backscatter, the pressure amplitudes and coherence values differ considerably between the two surfaces, and the differences increase with increasing receiver separation. For forward scatter the differences are much less. [Work supported by NORDA.]

1:20

HHH2. Variability of acoustic reflectivity over a manganese nodule field. F. N. Spiess and Marco M. P. Weydert (Marine Physical Laboratory of the Scripps Institution of Oceanography, University of California, San Diego, San Diego, CA 92152)

The backscattering from a manganese nodule field is studied. The theory predicts a dependence of reflectivity on acoustic wavelength, nodule shape, nodule size and number of nodules per unit area. First results from a recent deep tow expedition (ECHO Leg 01, June 1983) are presented. The variability of reflectivity with nodule coverage and size is discussed for each of several frequencies (4.5, 9, 28, 60, 110, 163 kHz). The acoustic data, obtained some 70 m above the deep sea floor, are compared with bottom photographs and box cores taken along the same path.

1:35

HHH3. Wind- and wave-induced acoustic noise at very low frequencies. A. C. Kibblewhite,¹⁾ K. C. Ewans, and D. C. Coup (Department of Physics, University of Auckland, Auckland, New Zealand)

Underwater ambient noise is known to be dependent on the wind. Several mechanisms have been proposed to explain the nature of the transfer of energy from the wind field to the acoustic noise field. Examples include wind and wave turbulence and nonlinear interactions between surface waves, which given rise to many phenomena not present in linear theory. This study examines these wind-related mechanisms from the low end of the acoustic spectrum. Data from a long-term study of ocean waves and the associated microseismic field recorded ashore has produced evidence helpful to the identification of processes active in the acoustic field at somewhat higher frequencies. The unique environment under which these recordings were made in New Zealand has contributed to the success of the study. Correlations with ambient noise recorded by a shallow water hydrophone were also made.¹⁾ Visiting Scientist, Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712-8029.

HHH4. High-frequency acoustic sea surface scatter correlated with wind and wave measurements. Steven O. McConnell (Applied Physics Laboratory, University of Washington, Seattle, WA 98105), Serhad Atakturk, and Kristina B. Katsaros (Department of Atmospheric Sciences, University of Washington, Seattle, WA 98195)

Backscatter and forward-scatter measurements were made at frequencies ranging from 20 to 50 kHz (principally 21 and 40 kHz) in Lake Washington. Simultaneous measurements were made of the wind speed, stress, and direction, and of the large and small scale wave height up to frequencies of 30 Hz. The wave height sensors penetrated the water surface at the approximate center of the area of ensonification, allowing a spatially as well as temporally simultaneous comparison of scattering levels with wave height. Preliminary results show the backscatter level to be about 10 dB less than in the open ocean for a given wind speed. This result may be due to differences in near-surface bubble size density distributions between salt and fresh water [Scott, *Deep-Sea Res.* 22, 653-657 (1975)].

2:05

HHH5. Ocean ambient noise at very low frequencies. Frederick D. Cotaras, Harold M. Merklinger, and Ian A. Fraser (Defense Research Establishment Atlantic, P. O. Box 1012, Dartmouth, Nova Scotia, Canada B2Y3Z7)

Ambient noise in the frequency range 1 to 20 Hz was measured at a number of sites in the North Atlantic. Freely drifting, surface-suspended hydrophones were used. A digital radio telemetry link was used to transmit the acoustic data to an attending ship. It is shown that the character of the noise in the 1-4-Hz range is different from that in the 6-20-Hz range. The distribution of 5-min averages of noise power in narrow frequency bands shows significant differences in standard deviation, skew, and kurtosis observed above and below 5 Hz. The high coherence of noise measured by two independent measurement systems 150-300 m apart horizontally gives strong evidence that the noise measured in the 1-5-Hz range was not flow, turbulence, or electrical noise related. There is an indication in the data that two noise mechanisms may be responsible for the noise in the 1-5-Hz range; at least one of these is wind speed related.

2:20

HHH6. High-frequency scattering from the sea surface. Suzanne T. McDaniel (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

Composite roughness theory provides one means of treating high-frequency scattering from the sea surface. In this method, the surface is artificially partitioned into two wavenumber regimes leading to results that are dependent on precisely how the surface is partitioned. In this paper, diffractive corrections to the geometrical optics solution of the Kirchhoff integral are derived. The correction terms, in this case, arise quite naturally from an asymptotic expansion of the scattering integral and are, hence, not dependent on the choice of an arbitrary parameter. Numerical results are obtained for a two-dimensional surface and compared with the results of composite roughness theory. [Research supported by NORDA.]

2:35

HHH7. Comparisons between sinusoidal boundary scattering theories and measured data. Diana F. McCammon (Applied Research Laboratory, The Pennsylvania State University, University Park, PA 16802)

An exact solution to scattering from a sinusoidal boundary has been presented by Holford [J. Acoust. Soc. Am. 70, 1116-1128 (1981)]. This solution, and the approximate solutions of Kirchhoff, Eckart, Brekhovskikh, and Rayleigh are contrasted with the measured data given by La-Casce and Tamarkin [J. Appl. Phys. 27 (2) (1956)] for three different surface roughnesses. Very good agreement is obtained by the Holford theory, especially in the location of reflection amplitude peaks and zeros.

HHH8. Application of the composite roughness model to bottom backscattering. Darrell R. Jackson (Applied Physics Laboratory, University of Washington, Seattle, WA 98105), Dale P. Winebrenner, and A. Ishimaru (Department of Electrical Engineering, University of Washington, Seattle, WA 98195)

Ocean bottom acoustic backscattering is modeled as an incoherent sum of intensities due to interface roughness and volume inhomogeneity. Rough interface scattering is treated by a version of the composite roughness model similar to that applied to the sea surface [S. T. McDaniel and A. D. Gorman, *J. Geophys. Res.* 87, 4127-4136 (1982)]. The volume scattering portion of the model employs the first-order multiple-scattering approximation including refraction and large-scale interface slopes, but neglects layering [A. N. Ivakin and Yu. P. Lysanov, *Sov. Phys. Acoust.* 27, 212-215 (1981)]. The model is compared with data presented at the Cincinnati meeting [J. J. Crisp, P. A. G. Thomas, A. M. Baird, and D. R. Jackson, *J. Acoust. Soc. Am. Suppl.* 1 73, S97 (1983)] using power-law fits to measured bottom relief spectra. The relatively slow fall-off of these spectra causes the composite roughness model to be sensitive to the large-scale cutoff wavenumber. [Work supported by ONR.]

3:05

HHH9. Spectra and limits of boundary waves at grazing incidence over rough surfaces. Gerald L. D'Spain,¹⁾ Emily H. Childs,¹⁾ and Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93940)

Previously we have presented measurements of the large, coherently scattered, boundary wave amplitude and dispersion for low-frequency grazing propagation over rigid planes with rigid hemispherical or hemicylindrical bosses. Recent laboratory studies with wedge corrugations and with spherical roughness elements, and for ranges up to $kR = 3000$, show that: (a) the growth of the boundary wave amplitude is self-limited; and (b) the spectrum of the boundary wave amplitude is a function of the slope and size of the wedge corrugations. [Research supported by the Office of Naval Research.]¹⁾ Ocean Acoustics Associates.

3:20

HHH10. Coherent response to a point source irradiating a rough plane. R. J. Lucas¹⁾ and V. Twersky (Department of Mathematics, University of Illinois, Chicago, IL 60680)

We consider a point source of sound exciting a planar rough surface, and write the coherent response as the Sommerfeld-Weyl integral in terms of the reflection coefficient (R) for correlated distributions of protuberances (bosses) on rigid or free base planes. The coefficient R and the associated angle-dependent impedance are determined by the ensemble-averaged multiple scattering amplitude (F) for one fixed boss of revolution, and F is specified by its single scattered value and the statistical-mechanics radial distribution function (p) for impenetrable disks. Low-frequency forms in terms of simple integrals of p delineate multipole-coupling and packing effects on propagation and attenuation (arising from incoherent scattering as well as absorption) as functions of the fraction (w) of base plane covered by bosses. Approximations of the response integral and corresponding plots are presented to exhibit the dependence on w and on other key parameters, and near-grazing aspects are emphasized.¹⁾ Visiting from Mathematics Dept., Loyola University, Chicago, IL.

3:35

HHH11. Model of high-frequency ambient noise in coastal waters. David Dunlap (Bendix Oceanics, 15825 Roxford Street, Sylmar, CA 91342)

A computer simulation of the directional dependence and level of intensity of high-frequency ambient noise in coastal waters has been developed. Many investigators have shown the directional dependence of surface generated noise. This paper extends that work to higher frequencies to simulate the background in which some active sonar systems must

operate. Contributions from thermal noise and biologically produced noise are also considered in the total levels. The surface wind noise component is modelled as a variable geometric pattern of point sources whose density may also be varied. Initial levels and bottom reflection coefficients are taken from the literature, and results are compared with measured data. The noise intensity is presented as a function of frequency, depth and vertical direction.

3:50

HHH12. Measurements of wind-generated noise levels in the Eastern Indian Ocean. D. J. Kewley and A. S. Burgess (Department of Defense, Weapons Systems Research Laboratory, Defense Research Centre Salisbury, South Australia)

Recordings of ambient sea noise made with a vertical linear hydrophone array (VLA) at five sites in the Eastern Indian Ocean have been analyzed for wind-speed dependence. Use is made of the vertical directional discrimination of the VLA to resolve noise generated at a distance and locally generated wind noise. The results are compared with data from similar experiments off the east coast of Australia [A. S. Burgess and D. J. Kewley, *J. Acoust. Soc. Am.* **73**, 201-210 (1983)].

4:05

HHH13. Observations of scattering at 300 kHz. Lloyd C. Huff and Robert G. Williams (NOAA, National Ocean Service, 6001 Executive Boulevard, Rockville, MD 20852)

Observations made with a modified Ametek-Straza model DCP4400/300 Doppler current profiler during an experiment to characterize the current measurement performance of the DCP4400/300 in a bottom-mounted, upward-looking configuration afforded the opportunity to observe narrow-beam backscattering over an interesting range of environmental conditions. CTD casts, water samples for particulate distributions, and wind velocity measurements were made concurrently with the acoustic observations. The functional forms of surface backscatter at

normal and 30° angles of incidence versus windspeed are presented. Profiles of volume scattering are shown in relation to the water column characteristics. The bottom scattering, as observed via a surface-bottom-surface path is controlled by surface roughness. Statistical distributions of the backscattering observations are compared with the Rayleighian probability density function. These observations clearly indicate the potential of utilizing the DCP4400/300 to gain useful information about sea surface conditions and about high-frequency backscattering in general.

4:20

HHH14. Sonar estimates of sea floor microroughness. T. K. Stanton (Marine Studies Center, University of Wisconsin, Madison, WI 53706)

This paper presents an analysis of the effects of microroughness of the ocean bottom on a sonar signal. The results best apply to features where the roughness amplitude is less than one-quarter of an acoustic wavelength such as with ripples, beds of rocks, and nodules. The shape of the probability density function (PDF) of the echo envelope is examined in terms of the rms roughness and a new parameter, the correlation area of the bottom. The area is equal to the product of the x and y correlation distances along the floor. The PDF is shown to be extremely sensitive to small changes in the roughness. Furthermore by determining the rms roughness from standard coherent reflection measurements, the correlation area may be extracted directly from the PDF. Thus both vertical as well as lateral information is obtainable from sonar data. The technique can be used to discriminate between different types of bottoms that may have the same roughness but different correlation areas, for example a floor with ripples versus one with rocks or nodules. The analysis combines two approaches: (1) A general statistical model employing the Rice PDF (S.O. Rice, in *Selected Papers on Noise and Stochastic Processes*, edited by N. Wax (Dover, New York, 1954), pp.133-294] and (2) a theory originated by Eckart [C. Eckart, *J. Acoust. Soc. Am.* **25**, 566-570 (1953)]. The analysis is applied to sonar data collected from the continental shelf near Cape Hatteras, North Carolina. The results are consistent with the known characteristics of the area.