

## High-intensity, focused ultrasonic fields

**Jensen, Leif Bjørnø**

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# PROGRAM OF

## The 115th Meeting of the Acoustical Society of America

Seattle Sheraton Hotel and Towers • Seattle, Washington • 16–20 May 1988

MONDAY EVENING, 16 MAY 1988

WEST BALLROOM, 7:00 TO 9:00 P.M.

### Tutorial Lecture

**Acoustic intensity technique.** Jiri Tichy (Pennsylvania State University, Graduate Program in Acoustics, Applied Research Laboratory, University Park, PA 16802)

Radiation and propagation of sound energy represents fundamental knowledge in all subfields of acoustics and vibration. The intensity technique was initially developed for sound power measurements but it can be used as a general tool for sound field investigations. This is possible by recent developments of signal processing and precision instrumentation technology which permit identification of detailed behavior of vibrating surfaces, mapping of sound fields in terms of sound pressure, particle velocity, complex acoustic intensity and energy densities, construction of wave fronts, and determination of phase and group velocities. The tutorial will cover the fundamentals of all aspects of the intensity technique. Basic theoretical relationships of sound energy generation and propagation will be discussed. The basics of the measurement technique, using two-microphone probes or pressure-velocity probes, and the appropriate dedicated instrumentation or more general computer use, will be discussed. The tutorial will briefly handle the precision of the measurements. A great deal of time will be devoted to examples of applications on radiation from simple and complex sources and their nearfields. The maps of energy propagation, sound pressure and particle velocity, as well as the formation of vortices, will be shown. Specific applications in many areas of acoustics will be summarized.

After the lecture, various companies will demonstrate the latest commercial equipment available to make acoustic intensity measurements.

TUESDAY MORNING, 17 MAY 1988

GRAND BALLROOM C, 8:15 TO 11:42 A.M.

### Session A. Speech Communication I: Analysis, Coding, and Synthesis

Peter Benson, Chairman

*ITT Defence Communication Division, 10060 Carroll Canyon Road, San Diego, California 92131*

Chairman's Introduction—8:15

#### Contributed Papers

8:17

**A1. An optimal class of generalized time-frequency representations for nonstationary signal analysis.** Yunxin Zhao, Les E. Atlas, and Robert J. Marks, II (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

In this paper, a class of generalized time-frequency representations (GTFR) that have both good time and good frequency characteristics for nonstationary signal analysis is presented. The basis of the approach used is that a time-frequency representation can, in some cases, be improved by allowing negative value of the spectrum. The representation (and its associated kernel) can then be optimized to enhance the peaks in frequency while still maintaining finite time support. The finite time support property has the significance of preserving onset times of signals and also providing clear representations of fast-changing spectral peaks. Experiments were performed on simulated data and real speech for comparison of the GTFR with the spectrogram and the pseudo-Wigner distribution. The results show distinct advantages of the GTFR. For example, onsets of transitions are clearer and, in the case of speech, close formant peaks are easier to distinguish. [Research supported by NSF and Boeing.]

8:32

**A2. Formant estimation from noisy voiced speech.** A. K. Krishnamurthy, J. Li, and R. L. Moses (Department of Electrical Engineering, The Ohio State University, 205 Dreese Laboratories, 2015 Neil Avenue, Columbus, OH 43210)

This paper considers the estimation for formant frequencies and bandwidths from voiced speech signals that are degraded with additive white noise. Each period of the speech signal is divided into the open and closed glottal phases using the electroglottograph signal. The speech signal in the closed phase is modeled as the sum of damped sinusoids, with each sinusoid corresponding to one formant. The analysis procedure thus leads to an estimate of the frequency, bandwidth, and energy of each formant. The results indicate that typically just three formants account for most of the energy in the closed phase. The noise-robustness properties of the algorithm are increased by using a total least-squares approach in obtaining the parameters and by using data from multiple consecutive closed phases. This work verifies and extends the results of S. Parthasarathy and D. Tufts [IEEE Trans. Acoust. Speech Signal Process. ASSP-35, 1241–1249 (1987)]. [Work supported by the OSU Seed grant program.]

**A3. Voiced speech model including source-tract interaction.** A. K. Krishnamurthy and J. Li (Department of Electrical Engineering, The Ohio State University, 205 Drees Laboratories, 2015 Neil Avenue, Columbus, OH 43210)

This paper considers the problem of estimating the parameters of a voiced speech model that includes the effects of source-tract interaction. Based on work by Ananthapadmanabha and Fant [Speech Commun. 1, 167-184 (1982)], the voiced speech signal is modeled as the output of a time-varying vocal tract filter excited by a parametrized glottal source waveform. The glottal source waveform [Fant *et al.*, STL/QPSR (1984)] models the main pulse shape of the glottal volume velocity and is described by four parameters. The effect of source-tract interaction is modeled by varying the frequency and bandwidth of the first formant in synchrony with the glottal source waveform. An analysis-by-synthesis approach is used to estimate the parameters of the model. Algorithms for estimating the parameters of the glottal source waveform and the vocal tract filter are described. Comparisons of the spectrum of the original and synthesized speech waveforms are presented. [Work supported by the OSU Seed grant program.]

**A4. Distribution of spectral errors with quantization of frequencies and bandwidths of LPC poles.** Bishnu S. Atal (Acoustics Research Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

In low bit rate speech coding systems, the LPC spectral information is often coded in terms of quantized partial correlations or line-spectral pairs. The distribution of spectral errors with such quantization has been studied in the past. However, not much is known about the distribution of spectral errors when the frequencies and bandwidths of LPC poles are quantized. The quantization of frequencies and bandwidths has important advantages because the perceptual sensitivity to quantization of these parameters can be estimated from data on just-noticeable differences. In this paper, results will be presented on the spectral error introduced by the quantization of frequencies and bandwidths of LPC poles as a function of the different number of predictor coefficients and the different number of bits used to represent the spectrum. The distribution of spectral errors using frequency-bandwidth quantization will be compared with other procedures based on quantization of partial correlations and line-spectral pairs, and relative strengths and weaknesses of these quantization schemes will be discussed.

**A7. On the semantic content of intelligibility test material on assessing signal processing algorithms.** P. Benson and J. Naylor (ITT Defense Communication Division, San Diego, CA 92131)

Signal processing algorithms can be used to enhance speech intelligibility by improving the detectability of important acoustic features. It should follow from this that other nonacoustic portions of the signal are a source of experimental error that should be controlled. Nevertheless, certain signal processing techniques produce reliable improvements in intelligibility only when that speech is meaningful. A series of intelligibility tests was carried out with co-channel speech, which combines the speech of two talkers into a single channel. This speech had been processed using the harmonic magnitude suppression (HMS) technique [Naylor and Boll, ICASSP (1987)] to diminish the effect of the louder talker. Intelligi-

**A5. A real-time frequency-shift decoder for the last transmission of Korean Airlines flight 007.** John D. Schlatter and Les E. Atlas (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

On 1 September 1983, Korean Airlines flight 007 was intercepted and destroyed by a Soviet jet fighter after straying into Soviet airspace. Although the flight recorder from KAL 007 was not recovered, its final transmissions were recorded at a Tokyo air traffic control center. The intelligibility of the KAL 007 transmissions is compromised by at least three types of distortion: broadband background noise, narrow-band noise tone, and frequency-shift distortion. This paper describes a DSP system that has been developed to counteract the frequency-shift distortion. The system employs Hilbert transform-based techniques to introduce a linear compensating frequency shift, the extent of which may be controlled manually by the user or automatically by a separate computer. Tests performed to date with the system on the KAL 007 transmissions have shown it to be a useful tool. Intelligibility improvements have led to independently verified advances in understanding several of KAL 007's final transmissions.

**A6. Large-band integrated vowel representations.** Marios Mantakas, Jean-Luc Schwartz, and Pierre Escudier (Laboratoire de la Communication Parlée, I.C.P.-Unité associée au C.N.R.S., E.N.S.E.R.G.-I.N.P.G., 46 avenue Félix-Viallet, 38031 Grenoble, France)

"Gross" vowel spectrum parameters for vowel classification are of interest to many researchers. In the present study, the "effective formant"  $F'2$ , estimated by the large-band spectral integration (LBI) model [Escudier *et al.*, *Acts of the French-Swedish Seminar*, Grenoble, France (1985)], in the classification of natural French front vowels /i/ vs /y/ and /e/ vs /ø/ (rounding opposition), is evaluated. In parallel, formants' measures on the same corpus are provided in order to compare with and interpret the results obtained using the LBI model. Classification performance is good in the case of /i/-/y/. The optimal spectral integration window is 2-2.5 Bark large (cf. with the 3-3.5 Bark "critical distance"). The model fails, however, in the /e/-/ø/ classification. Here, the second formant's frequency is by far winning. The model's weakness apparently resides in its last stage, namely peak estimation. In fact, recent results show that the LBI spectral representation can support the definition of form factors (other than peak position) for the classification of vowels.

### Break

bility testing using target speech of read PB sentences masking read four-word nonsense sentences revealed no differences between treated and untreated speech. Intelligibility using complete sentences as both masker and target showed significant differences in favor of the HMS treated speech. An analysis of signals and errors will be presented to show how listeners are integrating information from nonsignal sources to improve performance.

**A8. Preliminary study of multilevel peak-clipped and time-quantized speech.** Edward M. O'Brien and David J. Pogue (Bioengineering Program, Texas A&M University, College Station, TX 77843-3120)

Speech was amplified to a level of plus and minus 5 V for analog-to-digital conversion (sampling rate of 20 kHz). It was then quantized to five discrete levels. The 0-V level was used for speech within the squelch level (squelch was used to mask background noise). Positive speech waves above A volts were assigned a value of 5 V. Speech waves with peak values below this level were assigned a value of A volts. Negative speech waves were quantized to either  $-5$  V or  $-A$  volts similar to the processing of the positive speech waveforms. Processing was done with a PC. Two different A levels were tested on monosyllabic word lists. The first was at the level of the rms value of each word (average rms level across all words was 1.22 V). The second level was at a fixed value of 2 V. Preliminary results yielded a higher percentage word intelligibility. This type of speech processing has application to the development of a tactile hearing aid.

10:27

**A9. A microcomputer-based system for high-speed processing of speech materials using pitch-synchronous LPC analysis and synthesis.** Stephen J. Eady, B. Craig Dickson, Roy C. Snell (Centre for Speech Technology Research, University of Victoria, P.O. Box 1700, Victoria, British Columbia V8W 2Y2, Canada), and Melvyn R. Hunt (National Research Council of Canada, Building U-61, Montreal Road, Ottawa, Ontario K1A 0R6, Canada)

This paper describes a microcomputer-based system that uses a high-speed digital signal processor (TMS-32030) for digitizing and encoding speech materials. The system uses pitch-synchronous LPC (covariance method), in conjunction with a laryngograph, to analyze previously recorded speech materials [M. J. Hunt and C. E. Harvenberg, *Proceedings of the 12th International Congress on Acoustics*, A4-2 (1986)]. The speech and laryngograph signals are digitized simultaneously, and speech parameters ( $F_0$ , energy, formants, and bandwidths) are calculated in real time. The parameters are modified using an interactive editor, and real-time synthesis allows for auditory monitoring of the original and modified speech samples. The system is currently being used to process speech materials for a limited-vocabulary, word-concatenation synthesis system [Eady *et al.*, *Proc. ICASSP*, 1473-1476 (1987)], and also in the development of a demissyllable-based text-to-speech device. [Work supported by National Research Council Canada and Science Council of British Columbia.]

10:42

**A10. Adjusting syllable durations in a demissyllable speech synthesis system.** Suzanne C. Urbanczyk, Stephen J. Eady, and B. Craig Dickson (Centre for Speech Technology Research, University of Victoria, P.O. Box 1700, Victoria, British Columbia V8W 2Y2, Canada)

This paper describes the durational aspects of a speech synthesis system designed to generate words and sentences of English based on smaller synthetic speech units known as demissyllables [O. Fujimura, *J. Acoust. Soc. Am. Suppl.* 1 59, S55 (1976)]. Each syllable of a word is composed of an initial and a final demissyllable. The demissyllable inventory (totaling less than 900 units) is prerecorded and stored in LPC format. Demissyllables are linked to form syllables, which are then joined into words. Syllable duration is one of the more complicated features of word formation using this technique. Our strategy for producing syllables of appropriate length is to record the demissyllable units with relatively long durations and then to reduce the duration of the demissyllable components when they are joined together. Durational adjustment is accomplished using a spectral distance metric, which identifies regions of the syllable where the spectral components are least dynamic. The less dynamic regions can then be deleted to shorten a syllable, while maintaining perceptually relevant acoustic characteristics. This method is used to adjust syllable durations at both the word and sentence levels. [Work supported by Science Council of British Columbia.]

10:57

**A11. Delta streams for English synthesis.** Susan R. Hertz (Eloquent Technology, 254 Highgate Circle, Ithaca, NY 14850 and Phonetics Laboratory, Cornell University, Ithaca, NY 14853)

Although speech patterns are heavily influenced by the hierarchical linguistic structure of speech, synthesis by rule systems have generally been based on linear utterance representations. Delta is a new programming language that makes it easy to work with utterance representations containing multiple levels. Both higher level linguistic units, such as phrases, syllables, and phonemes, and lower level phonetic events, such as articulatory or formant targets and  $F_0$  trends, can be easily accommodated on separate interconnected "streams," with each unit equally accessible to the rules. While Delta can be used to test most any synthesis model for any language, this paper will show how Delta can be used to test a particular model for English. This model uses, among others, CV, syllable, nucleus, phoneme, formant, and duration streams, with formant transitions represented as duration tokens that are in effect invisible in other streams. The paper will justify the selection of streams and the unique way of handling formant transitions, demonstrating in Delta notation how the model leads to particularly straightforward rules for predicting English phoneme durations, formant values, and aspiration patterns.

11:12

**A12. A new design of a digital conference system.** Hans W. Gierlich (Institute for Communication Engineering, RWTH Aachen, Melatener Strasse 23, 5100 Aachen, West Germany)

Two main drawbacks make telephoning using a conference system troublesome: First, noise and reverberation produced at the site of the speaker are transmitted to the listener. Second, to achieve a sufficient speech volume at the listener's site, the necessary amplification for loud-speaker playback leads to instability (feedback) of the whole system. To suppress reverberation and noise, a speech controlled microphone arrangement is used. The implemented algorithm for speech detection is able to distinguish between noise and speech up to noise levels of 70 dB SPL. The noise signal may either be noiselike, impulselike, or sinusoidal. The detection algorithm needs a maximum time of 32 ms to detect a speech signal. To avoid instability, a combination of a special filter arrangement and a speech controlled amplification of microphone and loudspeaker signal is used. Two filters, having a transfer characteristic similar to a comb filter, are inserted into the transmitting and the receiving path. The transfer functions of the filters are inverse. With regard to the tone color, the filters were optimized, taking into account the special effects of signal processing used by the human ear. Hearing tests showed that speech intelligibility is not influenced by the filter arrangement.

11:27

**A13. The smoothed pseudo-Wigner distribution in speech analysis.** Labib M. Khadra (The University of Trondheim, Division of Telecommunications, Trondheim, Norway)

Recently, a great deal of interest has been shown in applying the Wigner-Ville distribution to obtain time-frequency energy representation of nonstationary signals. The utility of the Wigner distribution has proved useful in analyzing monocomponent signals. In the case of multicomponent signals, the Wigner distribution adds cross terms without any physical significance to the time-frequency distribution. The presence of cross terms obscures the actual spectral features of interest and makes the results very misleading. To apply the Wigner-Ville distribution to speech signals, it is essential to remove all cross terms not of interest. This problem may be solved by smoothing the Wigner-Ville distribution independently in time and frequency directions using the "smoothed" pseudo-Wigner estimator. This estimator possesses several advantages over the short time periodogram. This presentation will concentrate on the application of the smoothed pseudo-Wigner distribution (SPWD) to speech signals and will demonstrate the ability of the SPWD to improve the



quality of the time-frequency representation of speech signals. In particular, a comparison will be made between the spectrogram and the SPWD, and it will be shown how high-frequency spectral features may be easily

detected from an SPWD but are far less obvious in the spectrogram. Moreover, it will be shown that the SPWD is more appropriate for the analysis of formant structure.

TUESDAY MORNING, 17 MAY 1988

WEST BALLROOM A, 8:30 TO 11:45 A.M.

### Session B. Physical Acoustics I: Nonlinear Acoustics

Jacqueline Naze Tjøtta and Sigve Tjøtta, Cochairmen

*Applied Research Laboratories, The University of Texas at Austin, Austin, Texas 78713-8029 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway*

#### Contributed Papers

8:30

**B1. Ray solution for finite amplitude two-dimensional waves in a hard-walled rectangular waveguide.** Kun-tien Shu and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This paper describes a ray superposition theory for cumulative growth of nonlinear effects in a two-dimensional acoustic mode, based on decomposition of the mode into a pair of obliquely propagating, nonlinear planar waves. The mathematical foundation of the formulation is an earlier perturbation analysis of the reflection of a distorted planar wave obliquely incident on the boundary of an infinite half-space [Z. Qian, *Sci. Sin.* 25, 492-501 (1982)]. Based on the results of the analysis, each of the pair of rays forming the signal at a selected field point is traced back to its origin at the excitation. Each ray is described as a simple planar wave undergoing finite amplitude distortion that depends on the propagation distance along that ray between field and source points. This distance is the same for each ray at a specified field point, but differences in the excitation at the respective source points result in phase differences between the two rays. The overall signal is shown to be the same as a modal description of the propagation [H. C. Miao and J. H. Ginsberg, *J. Acoust. Soc. Am.* 80, 911-920 (1986)]. The ray solution explains an apparent paradox in the modal analysis, which indicated that although the signal can be resolved into a pair of planar waves, the distortion process is scaled only by the axial position along the waveguide. Conversely, the earlier solution provides validation for the superposition of rays, as well as for the linear reflection law. [Work supported by NSF and ONR.]

8:45

**B2. Similarity of a Fourier transform generalization of the Earnshaw solution for planar waves to an interacting wave model for finite amplitude effects in sound beams.** J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The Earnshaw solution for a finite amplitude planar wave, which displays amplitude dispersion, is valid for arbitrary excitation  $f(t)$  on a boundary. The form that results for small acoustic Mach numbers when  $f(t)$  is represented as the inverse of its Fourier transform  $F(\omega)$  may be considered to be a coordinate straining of the linearized signal, in which the transformation has the appearance of a Fredholm integral equation in the frequency domain. In the case of an acoustic planar wave, the phase speed of all frequency components is the same, independent of frequency. A generalization of that result is obtained if one considers the possibility that the cumulative growth effect is an arbitrary function of distance. Rather than being an abstraction, this form is shown to be analogous to an

earlier solution for finite amplitude sound beams [J. H. Ginsberg, H. C. Miao, and M. A. Foda, *J. Acoust. Soc. Am. Suppl.* 1 81, S25 (1987)]. The signal in that analysis was represented in terms of a spectrum of transverse wavenumbers, rather than frequencies, and the signal for each wavenumber was formed from interacting quasiconical waves, rather than a single planar wave. Nevertheless, the two problems share a common mathematical structure. A discussion of numerical algorithms for such models leads to some surprising observations regarding the implicit functional form of the Earnshaw solution. [Work supported by ONR.]

9:00

**B3. Scattering of sound by sound from two real beams.** Jarle Berntsen (Department of Informatics, The University of Bergen, 5007 Bergen, Norway), Jacqueline Naze Tjøtta, and Sigve Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway)

A theory for the nonlinear interaction between two sound beams produced by real sources in a lossless fluid was presented in a previous work [Naze Tjøtta and Tjøtta, *J. Acoust. Soc. Am.* 83, 487-495 (1988)]. A general solution of the governing equation in the quasilinear approximation, valid at any range, crossing angle, and frequency ratio, was obtained for prescribed boundary conditions. An asymptotic formula for the sum and difference frequency sound pressure was obtained at large distance from the sources. It relates the amplitude and directivity of the sound field in the farfield to the on-source conditions. In the present work, numerical results are presented for various types of sources (uniform piston, Gaussian source, focusing source). The influence of source geometry (separation, and intersection angles from  $0^\circ$  and  $90^\circ$ ) and frequency on the beam pattern of the nonlinearly generated sound is studied. Conditions are also given to determine when scattering of sound by sound can be observed. In the special case of thin Gaussian beams intersecting at a small angle, the results are compared with that presented by Darvennes and Hamilton [J. Acoust. Soc. Am. Suppl. 1 83, S4 (1988)] using the paraxial approximation. [Work supported by the IR&D program of ARL:UT, and VISTA/STRATOIL, Norway.]

9:15

**B4. Scattering of sound by sound from two Gaussian beams.** Corinne M. Darvennes and Mark F. Hamilton (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063)

The scattering of sound by sound from Gaussian beams that interact at small angles is investigated theoretically with a quasilinear solution of

the Khokhlov-Zabolotskaya nonlinear parabolic wave equation. The analytical solution, which is valid throughout the entire paraxial field, is a generalization of the result obtained for parametric receiving arrays by Hamilton, Naze Tjøtta, and Tjøtta [J. Acoust. Soc. Am. **82**, 311-318 (1987)]. Significant levels of scattered difference frequency sound are shown to exist outside the nonlinear interaction region. An asymptotic formula reveals that difference frequency sound is scattered in the approximate direction of  $\mathbf{k}_1 - \mathbf{k}_2$ , where  $\mathbf{k}_i$  is the wave vector associated with the direction and frequency of the  $i$ th primary beam. Computed propagation curves and beam patterns demonstrate the dependence of the scattered radiation on source separation, frequency ratio, interaction angle, and focusing. Results are also presented for the scattered sum frequency sound. Comparisons are made with the general asymptotic results presented by Berntsen, Naze Tjøtta, and Tjøtta [J. Acoust. Soc. Am. Suppl. **1** **83**, S4 (1988)], which are valid for arbitrary interaction angles, source separations, and amplitude distributions. [Work supported by ONR.]

9:30

**B5. Acoustical phase conjugation experiments: The generation of a reversed wave through three-wave mixing in a layer of stabilized microbubbles.** Steven G. Kargl and P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164-2814)

A phase-conjugate mirror is one that reverses an incident wave front so that it propagates back toward the source. Recent experiments [Kustov *et al.*, Sov. Phys. Acoust. **32**, 500-504 (1986)] indicate that wave-front reversal can be established through the interaction of a pump wave of frequency  $f_1$  with a probe wave of frequency  $f_2 < f_1$  that diverges from a point source. Three-wave (nonlinear) mixing occurred in a layer of freely rising gas bubbles in water so as to produce a reversed wave having a frequency  $f_3 = f_1 - f_2$ . The present research yields evidence of reversed wave generation resulting from three-wave mixing in a layer of stabilized microbubbles. One method of stabilization is to use the gas-filled micropores of a Nuclepore® polycarbonate membrane. Our experiments were carried out in a water tank of diameter 164 cm and with frequencies  $f_1$  and  $f_2 > 300$  kHz so as to avoid spurious boundary reflections. The previously predicted [P. L. Marston, J. Acoust. Soc. Am. Suppl. **1** **82**, 12-13 (1987)] longitudinal and transverse focal point shifts for the reversed wave are investigated. The signal for the reversed wave at frequency  $f_3$  is enhanced through the use of a background subtraction technique. [Work supported by ONR.]

9:45

**B6. Nonlinear equations of acoustics in inhomogeneous fluids.** Jacqueline Naze Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway), Edel Reiso (Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1063 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway), and Sigve Tjøtta (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713-8029 and Department of Mathematics, The University of Bergen, 5007 Bergen, Norway)

The propagation of finite amplitude sound waves produced by real sources in an inhomogeneous and thermoviscous fluid is considered. A governing nonlinear equation in the sound pressure amplitude is derived using the methods of singular perturbations. It consistently accounts for the effects of diffraction, dissipation, nonlinearity, and inhomogeneity, and represents a generalization of the parabolic equation valid for a homogeneous fluid (Khokhlov-Zabolotskaya-Kuznetsov equation) discussed in a previous work [Naze Tjøtta and Tjøtta, J. Acoust. Soc. Am. **69**, 1644-1652 (1980)]. The equation also applies to the case of sound beams produced by strongly curved sources, for example, focusing and defocusing sources. The relationship to the equations of classical ray theory is discussed. [Work supported by The Norwegian Research Council for Sciences and Humanities (NAVF), the IR&D program of ARL:UT, and VISTA/STATOIL, Norway.]

10:00

**B7. Weak foci in three-dimensional linear shock waves and the stability of real shock waves.** P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

A first approximation for identifying foci in weak shocks is to consider the focusing process for a linear discontinuity. This is sometimes known as the "acoustic shock" approximation. A case studied previously is the longitudinal cusp of "arête" located where three rays merge at the focus of a converging cylindrical wave; for finite amplitudes, the wave can become unstable with respect to the formation of a pair of shock-shocks in the focal region [B. Sturtevant and V. Kulkarny, J. Fluid Mech. **73**, 651-671 (1976)]. In the present research, catastrophe theory is used to identify wave-front shapes  $W(x,y)$  that propagate to produce novel acoustic foci with caustics spread out roughly transverse to the direction of propagation [P. L. Marston, J. Acoust. Soc. Am. **81**, 226-232 (1987) and *Proceedings of the APS 1987 Topical Conference on Shock Waves in Condensed Matter* (in press)]. The initial displacement of the wave front from the  $xy$  plane is given by  $W(x,y)$ . Linear waves with  $W = -(a_1x^2 + a_2y^2x + a_3y^2)$ , where the shape parameter  $a_2 \neq 0$ , propagate to produce transverse cusps described by a cubic cusp curve. A stronger focus, the hyperbolic umbilic, is produced if  $W = -[(ax^3 + 3\gamma^2x)/6 + a_1(x^2 + y^2)]$  with  $a_1 < 0$ . A focal section is produced at a distance  $z = -(2a_1)^{-1}$  from the  $xy$  plane. In this section, caustic lines form a V with an apex angle of  $2 \arctan(\gamma/\alpha)^{1/2}$ . Issues related to shock stability are noted. [Work supported by ONR.]

10:15

**B8. Calculation of the intensity and absorption of a finite-amplitude sound wave.** David T. Blackstock (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029 and Rochester Center for Biomedical Ultrasound, University of Rochester, Rochester, NY 14627)

An operational definition of absorption  $\alpha = -\nabla \cdot \mathbf{I} / 2I$ , where  $I$  is the sound intensity, was popular in the 1950s as a means of quantifying the increased attenuation of finite-amplitude ultrasonic beams. Note here that  $\alpha$  is the total absorption of the wave, not just the absorption of the fundamental component. Although the relation fell into disuse in the 1960s and 1970s, Carstensen *et al.* [Acustica **51**, 116-123 (1982)] recently revived it as a natural and useful tool to characterize attenuation in biological media. For a wave sinusoidal at the source, Carstensen *et al.* calculated  $\alpha$  and  $I$  from the Fourier series expression for the pressure of a finite-amplitude wave. Shown in this paper is an alternative calculation based on the time-domain version of the weak shock solution. Exact, closed-form expressions for  $I$  and  $\alpha$  are given for all distances from the source. The extension to pulses is indicated. [Work supported by NIH and ONR.]

10:30

**B9. Two-dimensional nonlinear wave propagation in an inhomogeneous elastic solid.** Hyun S. Kim and J. H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

This paper considers the two-dimensional problem of propagation of stress waves in an elastic half-space subjected to arbitrary excitation along a strip on the free surface. Inhomogeneity is taken to be a small-scale effect, in the form of small spatial fluctuations of the second-order elastic coefficients and density, where such variations are periodic functions of the vertical coordinate normal to the surface. General nonlinear equations of motion are derived in the Lagrangian form. Integral transform methods are used to solve the first- and second-order equations of motion, which are obtained from straightforward perturbation expansion of the displacement components. When the normal stress at the surface has a very slowly varying Gaussian distribution transversely to the strip, the asymptotic solutions are obtained in power series in terms of the small parameter for the Gaussian distribution. The longitudinal wave is shown to be the most dominant, although a shear wave does arise at the second

order of approximation. It is shown that secular solutions, which are usually associated with cumulatively growing nonlinear distortion, can also be produced by the inhomogeneity, if the periodicity of the exciting stress and inhomogeneity satisfy a certain relation. [Work supported by NSF.]

10:45

**B10. An investigation of the work output of a thermoacoustic prime mover.** Daniel C. Simard, Anthony A. Atchley, and Steven R. Baker (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

A thermoacoustic prime mover is a device that exploits an imposed temperature gradient to generate a work output. This work serves to offset dissipative losses and thus improves the quality factor of oscillations. The theory is developed using a quantitative expression for thermoacoustic work as derived by Swift *et al.* at Los Alamos National Laboratory [J. Acoust. Soc. Am. **78**, 767-776 (1985)]. An experiment is described in which the quality factor is measured as a function of temperature gradient. A modified version of the "Hofler" tube is used and the experiment is conducted in argon. The results indicate that the inverse of the quality factor is linear with temperature gradient and are in good agreement with theory. [Work supported by ONR.]

11:00

**B11. Low Prandtl number gas mixtures as a working fluid in a thermoacoustic refrigerator.** M. Suzalla, T. Hofler, and S. L. Garrett (Department of Physics, Naval Postgraduate School, Monterey, CA 93943)

Prandtl number (*Pr*) is the dimensionless ratio of kinematic viscosity to thermal diffusivity, and is about 0.7 for most ideal gases. This value can be lowered significantly by mixing two gas species having molecular weights that are very different, resulting in a minimum *Pr* of 0.22 for He-Xe mixtures. This can be used to minimize the nuisance effect of viscous shear losses for a thermoacoustic refrigerator as well as for other types of heat engines. The principle of thermoacoustic heat transport will be briefly discussed [J. Wheatley, T. Hofler, G. W. Swift, and A. Migliori, J. Acoust. Soc. Am. **74**, 153-170 (1983)]. However, changing the viscosity of the working fluid also changes the details of the acoustic velocity distribution, thereby modifying the basic thermoacoustic heat transport mechanism. Measurements indicate that this effect may be more important

than the simple reduction of viscous shear losses. [Work supported by the Office of Naval Research and the Office of Naval Technology.]

11:15

**B12. The effect of mean flow gradients, including shear, on the propagation of sound.** P. G. Vaidya (Mechanical and Materials Engineering Department, Washington State University, Pullman, WA 99164)

In the case of the perturbation analysis of nonlinear acoustics, it is customary to assume that the underlying undisturbed mean flow is uniform or nonexistent. When this assumption is removed, several interesting phenomena occur, which are of considerable practical significance. In this paper, three kinds of mean flow nonuniformities are analyzed. These are shear flows, axial gradients, and mean flows with nonzero divergence. It has been shown that these cases lead to sound attenuation, amplification, and instabilities, as well as the generation of fluctuating vorticity and the acousto-vortical waves.

11:30

**B13. Effect of stress on the temperature dependence of sound velocity.** John H. Cantrell (NASA Langley Research Center, Mail Stop 231, Hampton, VA 23665-5225)

The acoustic natural velocity for crystalline solids under stress is expressed directly as a function of the stress and the second- and third-order elastic constants referred to the zero stress state. The temperature dependence of the elastic constants is made explicit by use of the Helmholtz free energy for a system of quantized harmonic oscillators. The resulting expression is used to calculate the temperature derivatives of the sound velocity ( $dv/dT$ ) as a function of applied stress  $\sigma$  in the solid. It is found that  $(dv/dT)_\sigma^{-1} [(dv/dT)_\sigma - (dv/dT)_0] = K\sigma$ , where the subscripted  $\sigma$  denotes evaluation at stress  $\sigma$  and the subscripted zero denotes evaluation at zero stress. The constant  $K$  is found to depend explicitly on the third-, fourth-, and fifth-order elastic constants of the crystal. A central-force nearest-neighbor lattice model is used to calculate the fourth- and fifth-order constants from measured values of the acoustic nonlinearity parameters. For these elastic constant data,  $K$  (for stress applied perpendicular to the longitudinal wave propagation direction in isotropic solids) is calculated to be  $1.8 \times 10^{-3} \text{ MPa}^{-1}$  for pure aluminum and  $6.9 \times 10^{-4} \text{ MPa}^{-1}$  for copper. These results are in good agreement with experimental values obtained for aluminum and copper alloys [K. Salama and C. K. Ling, J. Appl. Phys. **51**, 1505 (1980)].

TUESDAY MORNING, 17 MAY 1988

EAST BALLROOM A, 8:30 TO 11:35 A.M.

### Session C. Structural Acoustics and Vibration I: Special Session on Response of Coupled Dynamic Systems

Gideon Maidanik, Chairman

David Taylor Research Center, Code 1902, Bethesda, Maryland 20084

Chairman's Introduction—8:30

Invited Papers

8:40

**C1. Modeling structural acoustics for active control.** A. H. von Flotow, D. W. Miller, and D. J. Pines (Department of Aeronautics and Astronautics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Active techniques are beginning to be considered for the control of structure-borne sound. Expertise from many fields is being brought to bear upon this problem, including experience in active control of the acoustic

response of ducts and air volumes, and the active control of flexible spacecraft. High-fidelity modeling is a necessary prerequisite to high-performance active control. This paper briefly reviews the past approaches to the modeling of structure-borne sound and approaches taken for their verification. The main body of the paper explains a recently developed approach to such modeling, applicable to a restricted class of structures. The paper closes with the description of two techniques developed for the design of active control systems for structure-borne sound, and describes preliminary laboratory experimental efforts.

9:05

**C2. Response of coupled one-dimensional dynamic systems.** J. Dickey, L. J. Maga, and G. Maidanik (David Taylor Research Center, Annapolis, MD 21402)

Attention is focused on a complex of one-dimensional dynamic systems. The extent of a dynamic system is defined in terms of two terminal positions, one at each of two junctions. A terminal vector per junction can thus be constructed. The propagation in a dynamic system is defined in terms of two propagation functions each describing the propagation toward one or the other junction. A diagonal propagation matrix per junction can thus be constructed. The coupling among the dynamic systems is defined in terms of two junction matrices. The external drive at a position on a given dynamic system is defined in terms of the responses that it initiates toward one junction and toward the other junction. With this model and definitions of a complex of coupled one-dimensional dynamic systems one may appropriately derive the various component matrices of the impulse response matrix. The matrix yields the response of the complex to the appropriately defined external drive vector. The derivation is explained and discussed. The basic (exponential-type) propagation forms and the nonbasic (e.g., Bessel functional-type) propagation forms are contrasted. The utility of the formalism is briefly cited and discussed.

9:30

**C3. Structural power flow analysis of coupled structures.** J. M. Cuschieri (Center for Acoustics and Vibration, Department of Ocean Engineering, Florida Atlantic University, Boca Raton, FL 33431)

Structural power flow techniques are an effective tool for the dynamic analysis of complex coupled structures, especially those types of coupled structures that are somewhat repetitive. Using the structural power flow method, the response of a structure is analyzed by tracking the flow of the vibrational power through the coupled structures, from the location of the excitation. An advantage of the power flow technique is that, if one component of the structure is modified, the analysis need only be repeated for the modified structural component, and the reevaluation of the coupling expressions. The use of the structural power flow method is demonstrated for beamlike and platelike structures including an analysis of the influence of structural and excitation parameters on the response of the coupled structures. Power flow methods can also be used to analyze more than one type of wave motion, which is useful in cases where incident wave power is scattered into other wave types near structural discontinuities. Additionally, experimental results for structural power flow are discussed. [Work supported by NASA.]

9:55

**C4. Coupled systems of anisotropic layers; A group formulation.** Michael Schoenberg (Schlumberger-Doll Research, Old Quarry Road, Ridgefield, CT 06877-4108) and Francis Muir (Department of Geophysics, Stanford University, Stanford, CA 94305-2215)

A matrix formalism allows for the simple calculation of the anisotropic, homogeneous medium equivalent to a stationary distribution of thin layers in welded contact. Each layer itself may be an elastic anisotropic medium. The calculated medium is equivalent to the layered medium in the long wavelength limit. The physical properties of any constituent of the heterogeneous system can be shown to be transformable to an element of a commutative group. Adding group elements  $G_A$  and  $G_B$  (corresponding to thin layers of constituents A and B, respectively) gives the group element for the homogeneous medium equivalent to the interleaved layers of A and B. The group formulation, which is essentially a statement of conservation laws applicable to layered media, enables us to "decompose" an anisotropic medium into several constituent sets of layers by successive additions of inverse elements, i.e., subtractions. If, after each subtraction, the remaining group element corresponds to a stable anisotropic medium, a valid decomposition is obtained. Within the group of all anisotropic constituents there are nests of subgroups, each subgroup corresponding to a symmetry system more restrictive than the most general triclinic anisotropy. Given the symmetry systems to which the constituents belong, the subgroup structure reveals, *a priori*, the symmetry of the equivalent medium.

### Contributed Papers

10:20

**C5. Dominant parameters of weakly coupled systems.** Takeru Igusa (Department of Civil Engineering, Northwestern University, Evanston, IL 60208)

A modal analysis framework was developed to identify the dominant characteristics of weakly coupled systems using perturbation analysis and mode synthesis techniques. Modal parameters were obtained to quantify four types of characteristics—tuning, coupling, nonclassical damping,

and spatial interaction. These parameters were subsequently used to determine the response of the system to broadband excitation. It was found that the parameters could be used to isolate certain modes of the system with dominant effects on the response. The response characteristics were also examined in the high-frequency range by extending asymptotic modal analysis [E. H. Dowell and Y. Kubota, *J. Appl. Mech.* **52**, 949-957 (1985)] to include the asymptotic properties of the coupled system parameters. The results were used to illustrate applications in design of isolation mounts in the low-frequency range, and in providing insight into statistical energy analysis in the high-frequency range. The work is an extension of the dynamic analysis of lumped mass secondary systems subjected to low-frequency excitation [T. Igusa and A. Der Kiureghian, *J. Eng. Mech.* **101**, 20-41 (1985)].

10:35

**C6. Effect of foundation design and resilient mount positions on structure-borne noise transmission.** Peter K. Kasper and Russel D. Miller (NKF Engineering, Inc., 12200 Sunrise Valley Drive, Reston, VA 22091)

A typical structure-borne noise control configuration consists of a vibrating machine, a set of resilient isolation mounts, a foundation supporting structure, and a building floor or deck. If the floor is represented by an infinite flat plate and the foundation is comprised of structural beam elements, the acoustic power transmitted to the floor can be calculated in exact fashion using the dynamic direct stiffness technique. This analysis approach involves the representation of beam segment properties in terms of dynamic frequency-dependent stiffness coefficients. The solution of dynamic displacement response to a harmonic input force parallels the solution of static deflections to an applied constant load. Calculations of transmitted acoustic power were made for a range of foundation structural parameters and resilient mounting configurations. The results indicate the sensitivity to such design parameters on structure-borne noise transmission.

10:50

**C7. Fluid-loaded constrained-layer analysis by exact elasticity theory.** Laurene V. Fausett (Department of Mathematical Sciences, Florida Institute of Technology, 150 W. University Boulevard, Melbourne, FL 32901) and Pieter S. Dubbelday (Underwater Sound Reference Detachment, Naval Research Laboratory, P.O. Box 568337, Orlando, FL 32856-8337)

The constrained-layer damping technique is based on the attachment of an absorbing viscoelastic layer and a stiff constraining layer to the surface of a metal structure to be damped. Starting from the model by F.

M. Kerwin [*J. Acoust. Soc. Am.* **31**, 952-962 (1959)], based on thin-plate theory for the base plate, a hybrid model of a fluid-loaded plate was developed whereby only the base plate is described by exact elasticity theory [P.S. Dubbelday, *J. Acoust. Soc. Am. Suppl.* **1 80**, S121 (1986)]. A description based on exact elasticity theory for all three layers, without fluid loading, was also formulated [D. W. Fausett, L. V. Fausett, and P. S. Dubbelday, *J. Acoust. Soc. Am. Suppl.* **1 80**, S121 (1986)]. The present study further explores the interaction of viscoelastic damping and fluid loading by a description according to exact elasticity theory of all three layers, combined with fluid loading. For flexural waves, the model shows reasonable agreement with the earlier hybrid model at intermediate frequencies, but discrepancies occur at lower and higher frequencies. These are discussed, and possible mechanisms for the behavior are brought forward. [Work supported by ONR.]

11:05

**C8. General relationships between acoustic impedance and modal density.** Jerome E. Manning (Cambridge Collaborative, Inc., Cambridge, MA 02138)

A general formulation of the acoustic impedance is set out using modal analysis. It is shown that the average acoustic resistance can be generally expressed in terms of the modal density of the acoustic space, the bulk compliance of the space, and the average joint acceptance. Examples are given for both simple and complex sources. Although the work presented offers no previously unknown solutions to acoustic radiation problems, it provides a very convenient engineering result that can be used in statistical energy analysis and other related acoustic analysis techniques.

11:20

**C9. An analytic model of shear and moment interaction of reinforcing ribs with cylindrical shells for acoustic radiation.** Courtney B. Burroughs (Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, PA 16804) and Kenneth J. Becker (Department of the Navy, Naval Sea Systems Command, Washington, DC 20362-5101)

In response to vibratory motion of a cylindrical shell, reactive forces and moments are generated by reinforcing ribs attached to the interior surface of the shell. These reactive forces and moments modify the shell motion and thereby the farfield acoustic radiation from the shell. Equations governing the motion of a rib, modeled as a circular ring, are derived that include normal and circumferential shear forces, and moments applied to the outer edge of the ring where the ring is attached to the inner surface of the shell. Approximate solutions for the impedances of the ring at the outer edge are obtained and used in equations of motion for a point-excited, fluid-loaded, circular cylindrical shell reinforced by periodically spaced ribs.

TUESDAY MORNING, 17 MAY 1988

WEST BALLROOM B, 8:30 TO 11:45 A.M.

## Session D. Underwater Acoustics I: Stochastic Volume and Boundary Scattering I

Terry Ewart, Chairman

*Applied Physics Laboratory HN-10, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105*

Chairman's Introduction—8:30

### Invited Papers

8:35

**D1. Acoustic volume scattering in the ocean—Approaching maturity.** B. J. Uscinski (Department of Applied Mathematics and Theoretical Physics, University of Cambridge, Cambridge CB3 9EW, England)

Until recently, the approximations of Born and Rytov were the standard methods used to describe the intensity fluctuations that arise when a sound wave is scattered in the body of the ocean. Since these methods

are valid only for very small fluctuations, it is not surprising that agreement with experiment was often very bad. Over the last few years new analytical expressions for the moments of the acoustic field, valid even for large intensity fluctuations, have been derived on the basis of the parabolic wave equation. These results have been very successful in explaining experiment observations and have also proved to be a powerful tool in furthering our understanding of fluctuation phenomena. From this point of view, acoustic volume scattering can be regarded as a problem for which some answers are now found. The expressions are for the second moment, or complex coherence, and the fourth moment, which gives the intensity fluctuation spectrum. The same forms have been derived by several independent methods and exist in the general case when a deterministic sound-speed profile leads to curved ray paths. The analytical intensity spectrum has been tested by comparison with accurate numerical solutions of the fourth moment equation and the agreement is excellent even for very strong scatter. The intensity fluctuation spectra observed in the experiments at Cobb Seamount are well described by their new fourth moment expressions, where Rytov's approximation had proved hopelessly inadequate. It is strongly recommended that small amplitude approximations, such as that of Rytov, should no longer be used to describe multiple scatter observations when the newer theory, tested and valid in these conditions, is now available.

9:00

**D2. The numerical approach to understanding rough surface scattering.** Eric I. Thorsos (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195)

Numerical methods for solving simplified rough surface scattering problems "exactly" provide new insights into the rough surface scattering problem. Exact solutions for 1-D surfaces with the pressure release (Dirichlet) boundary condition are found with integral equation and Monte Carlo techniques. These studies provide a better understanding of the validity of the two basic approximations used in rough surface scattering: the Kirchhoff approximation and first-order perturbation theory. From this work, a resolution can be given for the difference in the Kirchhoff and first-order perturbation theory predictions at small surface heights and at very small surface slopes, where both approximations have generally been considered valid. Studies with the Kirchhoff approximation show that shadowing corrections based on geometric optics lead to inaccurate predictions. This work indicates the need for tractable multiple scattering theory for low grazing angle applications. Comparisons with exact results show that "phase perturbation" theory predictions account for a significant part of the multiple scattering associated with shadowing. In addition to better understanding theory with numerical techniques, the accuracy of approximate numerical methods can also be examined. Attempts are now being made to include rough surface scattering in marching solutions of the parabolic wave equation (PE). Methods of verifying the accuracy of marching solutions using exact and PE integral equations will be briefly discussed.

9:25

**D3. A tutorial review of the role of scattering theory in detection and estimation of sonar signals.** Arthur B. Baggeroer (Departments of Ocean and Electrical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139), T. Ewart (Applied Physics Laboratory, University of Washington, Seattle, WA 98105), and John McCoy (Catholic University, Washington, DC 20064)

There are many theories that describe time and Doppler spreads, coherences and intensities, and higher-order probability densities of scattered signals. The sonar signal processor concerned with detection and estimation applications must use parts of these theories to design systems and analyze their performance. This talk will give a tutorial review of how scattering theories are used by the sonar signal processor.

9:50

**D4. The statistics of scattered waves.** E. Jakeman (Royal Signals and Radar Establishment, Malvern, Worcsire WR14 3PS, England)

The statistical properties of waves scattered by random media are a subject of long-standing interest both in the context of remote sensing and noise limitation of system performance. It is usually agreed that when many independent scattering centers contribute to the amplitude at the detector, then the statistics will be Gaussian as a consequence of the central limit theorem. There are many situations where this is not the case, however, and it is necessary to adopt non-Gaussian noise methods. In the past, these have often been chosen on the *ad hoc* basis of goodness of fit to limited data sets, rather than being well founded on physical insight and a clear understanding of the scattering mechanisms involved. More recently, it has been conjectured that hierarchical behavior within the scattering medium, which is often observed in nature, could give rise to rather characteristic non-Gaussian fluctuations. Supporting evidence from optical scattering experiments, from microwave sea echo data, and from certain exactly solvable configurations has encouraged the development of the  $K$ -distribution model. A general review of this area will be presented.

10:15

**D5. Comments regarding the two-scale expansion for the fourth moment of waves propagating in random media.** R. S. Patton and J. L. Codona (AT&T Bell Laboratories, Whippany, NJ 07981)

Over recent years, the method of two-scale expansions has been applied with a great deal of success to the problem of the fourth moment of waves propagating in random media. Specifically, the method yields a remarkably simple expression for the two-point intensity correlation that is valid in both strong and weak scattering and both high and low spatial frequencies. Unfortunately, of the published derivations, there is no clear presentation of the physical content of the method nor a straightforward approach for finding either corrections to the approximation or generalizations for treating waves in arbitrary refracting media. The clearest approach would appear to be based on standard configuration-space path integrals [B. J. Uscinski, C. Macaskill, and M. Spivack, *J. Sound Vib.* **106**, 509–528 (1986)]; however, that derivation appears to have (at the minimum) a different physical content from the other, moment equation based, derivations. The nature of the approximation is discussed in the context of both configuration-space and phase-space path integrals. The two-scale approximation is found to be a simple interpolating formula between a high spatial-frequency approximation (found by considering high-frequency asymptotics in phase space) and the low spatial-frequency behavior from the standard Born approximation. Looking at the approximation from this perspective lends both physical insight and simplifies the derivation to the point where it becomes quite simple to write expressions for refracting environments and correction terms.

10:30

**D6. Non-Kirchhoff elastic wave scattering from rough interfaces.** Henrik Schmidt (Massachusetts Institute of Technology, Cambridge, MA 02139) and W. A. Kuperman (Naval Research Laboratory, Washington, DC 20375)

An earlier developed solution technique for seismoacoustic scattering by stochastically rough interfaces [W. A. Kuperman and H. Schmidt, *J. Acoust. Soc. Am.* **79**, 1767 (1986)] has been modified to treat the scattering without applying the Kirchhoff approximation. This is accomplished by including the rotation of the boundary conditions in the perturbation formulation and by numerically evaluating the scattering integrals, allowing the roughness correlation lengths to be finite. It is demonstrated that the effect of the scattering on the mean (coherent) field can be accounted for in a self-consistent manner by solving the unperturbed problem with modified boundary conditions. As was the case for the Kirchhoff approximation, the non-Kirchhoff formulation has been implemented in the seismoacoustic SAFARI code [H. Schmidt and F. B. Jensen, *J. Acoust. Soc. Am.* **77**, 813 (1985)], allowing for simultaneous treatment of multiple rough interfaces. The model is applied to analyze the effect of scattering from a rough shallow water sea bed as well as from an Arctic ice canopy. By comparing these results to the Kirchhoff results for the same problems, it is demonstrated that the non-Kirchhoff theory predicts significantly higher losses, in particular for acoustic wavelengths of the same order of magnitude as the roughness correlation length. [Work supported by NRL.]

10:45

**D7. Kirchhoff approximation for scattering from a rough fluid–solid interface.** Dalcio K. Dacol (U.S. Naval Research Laboratory, Code 5160, Washington, DC 20375-5000)

A general prescription for the Kirchhoff approximation for wave scattering from a penetrable surface is implemented for the case of acoustic scattering from a rough fluid–elastic solid interface. A simpler version developed from this prescription produces results that compare favorably with exact numerical results for the case of simple periodic fluid–solid interfaces. This approximation is well suited to describe sound scattering from a randomly rough fluid–solid interface. In this case, the results obtained using this approximation are compared with those from a previous-

ly developed renormalized perturbation theory. Overall, this approximation is satisfactory but, since it is a single scattering theory, it fails to incorporate multiple scattering effects that are small in its domain of validity.

11:00

**D8. Rough surface scattering using the phase-perturbation technique.** Shira L. Broschat (Applied Physics Laboratory and Department of Electrical Engineering, University of Washington, Seattle, WA 98195), Eric I. Thorsos (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195), and Akira Ishimaru (Department of Electrical Engineering, University of Washington, Seattle, WA 98195)

There is a need for a solution to the problem of wave scattering from rough surfaces that is accurate when both the classical field perturbation and Kirchhoff (physical optics) approximations are not. In this work, the validity of the phase perturbation technique is examined for a region in parameter space when the two classical solutions fail. Numerical results for the phase perturbation scattering strength are compared with exact numerical results for one-dimensional pressure-release surfaces having a Gaussian roughness spectrum and Gaussian height and slope distributions. Exact results are obtained using a Monte Carlo technique. It is found that in the region considered the phase perturbation results agree with the exact results over all scattered angles away from low grazing for a fixed angle of incidence. Furthermore, in many cases it is found that exchanging incident and scattered angles in the phase perturbation equations gives an alternate phase perturbation solution whose numerical results are in excellent agreement with exact results. [Work supported by ONR.]

11:15

**D9. Efficient, high-frequency, parabolic-equation modeling with surface scattering loss.** Martha E. M. Head (Naval Oceanographic Office, Bay St. Louis, NSTL, MS 39522–5001), Eleanor S. Holmes, Lewis B. Dozier (Science Applications International Corporation, 1710 Goodridge Drive, McLean, VA 22102), and W. Jobst (Naval Oceanographic Office, Bay St. Louis, NSTL, MS 39522-5001)

Solving the parabolic approximation to the acoustic wave equation by the split-step algorithm is one of the principal methods for estimating the acoustic-pressure field in a range-dependent underwater environment. The algorithm is computationally intensive and is usually practical only at low frequencies. A modification to the split-step algorithm is provided (the calculation-frequency method), which allows rapid calculation of transmission loss at high frequency, includes surface and bottom loss, and provides for the redistribution of surface-scattered energy in wavenumber space.

11:30

**D10. Integral equations for the rough surface scattering amplitude.** John A. DeSanto (Department of Mathematics, Colorado School of Mines, Golden, CO 80401)

The rough surface scattering amplitude can be written as a surface integral over the boundary values of the field and its normal derivative on the surface. A general procedure is described to derive Lippmann–Schwinger integral equations in  $k$  space on these scattering amplitudes for Dirichlet, Neumann, and interface or transmission problems. The procedure is valid for arbitrary incident fields. It yields a multiple scattering description of the scattering process in terms of a fundamental surface interaction function that describes the scattering primarily as a phase modulation due to the roughness. Simple examples of single and double scattering for a one-dimensional surface will be presented. Also included are comments on the diffraction and scattering problems that arise when treating unbound surfaces. [Work supported by ARO.]

**Session E. Architectural Acoustics I and Noise I: Interior Noise of Aircraft**

Richard J. Peppin, Cochairman  
*Scantek, Inc., 51 Monroe Street, Rockville, Maryland 20850*

Dana S. Houglan, Cochairman  
*David L. Adams Associates, Inc., 1701 Boulder Street, Denver, Colorado 80211*

Chairman's Introduction—9:00

*Invited Papers*

9:05

**E1. The architectural space: A comparison of acoustical criteria used to assess aircraft noise in rooms and aircraft interiors.** Dana S. Houglan (David L. Adams Associates, Inc., 1701 Boulder Street, Denver, CO 80211)

The aircraft as a sound source is viewed in different ways depending upon the location of the receiver of concern. Although, often, the same source is dealt with, there are definite differences between the impact on a building occupant near an airport and the crew or passengers within the actual aircraft. Studies of both receivers use similar primary criteria such as speech intelligibility, fatigue, annoyance, and hearing loss. The criteria differ in relative importance, length of exposure, and perception of value. The application of secondary criteria used in noise control studies to quantitatively express attenuation through the exterior shell, acceptable background levels, and room effect, vary between the two fields of study. The comparison of similarities and differences in existing, commonly used criteria for these two fields of study is explored. Suggestions for areas of further study and correlation between the two fields of study are investigated.

9:30

**E2. Aircraft interior noise research at the NASA Langley Research Center.** Clemans A. Powell and Richard J. Silcox (NASA Langley Research Center, Mail Stop 463, Hampton, VA 23665-5225)

The Structural Acoustics Branch at the NASA Langley Research Center has been a major contributor to the field of aircraft interior noise research for the past 15 years. In addition to contributions from its in-house program, the branch has supported many contributions from other organizations through an active grant and contract research program. Although the current emphasis of the program is on advanced turboprop airplanes and helicopters, the goal has always been to develop and improve interior noise prediction methodology. This allows for the incorporation of appropriate control measures in new aircraft at the design stage rather than through the use of add-on acoustical treatments with potentially large weight penalties and reduced effectiveness. The proposed presentation will concentrate on some of the more recent basic research activities concerned with the transmission of noise into aircraft interiors through both airborne and structureborne paths. These include modeling of the structural response, the interior acoustic response, energy flow techniques, and acoustical-structural coupling. In addition, some results of recent activities in active noise control will be presented. Although the emphasis of the presentation will be on research conducted in-house, some research conducted under contracts and grants that is closely connected to in-house activities will also be discussed.

9:55

**E3. Interior noise characteristics of rotorcraft.** Charles R. Cox (Bell Helicopter Textron, MD-11, P. O. Box 482, Fort Worth, TX 76101)

The major noise sources and weight-efficient means for noise control in different size, single-rotor helicopters were determined by numerous measurements. Noise inside helicopters is characterized by numerous high-frequency tones during all flight modes and by broadband wind noise at cruise airspeeds. The tones, emanating primarily from the main transmission, the engine gearbox, and accessories, are the most objectionable and are present in the cabin of a helicopter at much higher levels than in the cabins of fixed-wing aircraft. Because of this characteristic, certain criteria used to determine acceptability, particularly speech intelligibility, are found inappropriate when applied to helicopter interior noise. Noise control is realized by design changes, e.g., source reduction and sound transmission path interruption, and by tailored soundproofing treatments. While some of these design changes are generic, most are unique to a particular helicopter model. This reflects the multiple



structural designs concepts, the variety of drive system mounting configurations, and extensive use of composite materials. Emerging challenges involve the noise control of all composite rotorcraft airframes and of the new tilt rotor aircraft.

10:20

**E4. Advanced acoustic test methods used in aircraft interior acoustic design.** Mark A. Lang (McDonnell Douglas Corporation, Douglas Aircraft Co., MC 36-80, Long Beach, CA 90846)

Advanced measurement techniques, particularly sound intensity measurement, are becoming an important design and diagnostic tool in aircraft design. The utilization of these tools in aircraft measurement presents unique challenges that must be overcome to obtain meaningful data. Once the data are obtained, careful analysis must be utilized in order to interpret the measurement results. A working group under SAE Committee A-21 is currently drafting a document intended to provide guidelines on sound intensity measurement under flight conditions. This document will be discussed in the light of some recent sound intensity measurements made in the Douglas Aircraft Fuselage Acoustic Research Facility as well as under flight conditions to highlight some of the measurement challenges involved in this type of measurement.

10:45

**E5. Test procedure for improved data quality in aircraft interior noise measurements.** Curtis I. Holmer (Noise Control Technology, Inc., 2440 Freetown Drive, Reston, VA 22091)

Vehicle interior noise measurements, and especially aircraft measurements, present a test environment that is inimical to considered data gathering due to the expense of operating time and the fact that the vehicle characteristics change during the measurement. A method for dealing with this is to construct a measurement procedure that provides for independent evaluation of data quality after the measurement is complete. One such method relies on the use of the relationship between the integral of intensity on a closed surface and the power produced or consumed within. Performing the measurement to permit estimation of this integral and an independent estimate of the power provides the opportunity to check data validity. This paper explores the approach in detail and examines the resulting data for a series of in-flight measurements. The measurements were performed on the Gulfstream IV aircraft as part of a program to define noise source areas. Value of the technique is explored as a function of several parameters including amount of absorption within the closed surface, reactivity of the measurement volume, and other significant parameters. [Work supported in part by Gulfstream Aerospace Corporation, Savannah, GA.]

11:10

**E6. Intensity measurements for interior noise control.** Robert L. Cohen (Boeing Commercial Airplanes, P. O. Box 3707, M/S 01-41, Seattle, WA 98124)

Typical noise sources, purposes of making intensity measurements, and difficulties in making useful intensity measurements in jet aircraft interiors are surveyed. Primary noise sources contributing to cabin interior noise are boundary layer, air conditioning, engines, and machinery. Transmission paths include both air and structure. Primary applications of intensity measurements in evaluating and controlling the effects of these sources on cabin noise include (1) locating and ranking sources for noise reduction; (2) optimizing acoustic treatment for minimizing weight; (3) providing data in support of cabin noise predictions related to planned changes in structure or acoustic treatment; and (4) transmission loss measurements for evaluating and improving sidewall panel acoustic effectiveness. Measurement difficulties include limited flight time, requirements for sampling detail in unknown gradients across each surface, unknown surface absorption, and unknown background noise levels interfering with the measurement of surface radiation, determining the accuracy of each measurement, and interpreting the data especially as a function of data quality.

### *Contributed Paper*

11:35

**E7. Using reciprocity to predict aircraft interior noise due to multiple correlated input forces.** Istvan L. Ver (BBN Laboratories Incorporated, 10 Moulton Street, Cambridge, MA 02238), William H. Mayes (Aircraft Noise Reduction Division, NASA Langley Research Center, Hampton, VA 23665), and Michael C. McGary (Boeing Commercial Airplanes, Seattle, WA 98124)

This paper reports on how to utilize reciprocity to predict aircraft

cabin noise due to multiple correlated point forces, such as those acting on engine mounting points. The transfer functions are determined by a reciprocal experiment by measuring, at the force excitation points, the magnitude and phase of the vibration response due to the operation of a point sound source located in the cabin. The experiments reported were carried out on the fuselage of a light aircraft. The methodology, the experimental apparatus, the software, and the test results will be presented and the advantages of the reciprocity method discussed. [Work supported by NASA Langley Research Center.]

**Session F. Education in Acoustics I: Undergraduate Student Projects (Poster Session)**

Les Atlas, Chairman

*Department of Electrical Engineering, FT-10, University of Washington, Seattle, Washington 98195***Contributed Papers**

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

**F1. Perceptual evidence for anticipatory assimilation of adjacent stops.**

Jeffrey W. Chan and Amy Dolcourt-McElroy (University of California, Berkeley, 1777 Euclid Avenue, Berkeley, CA 94709)

Prior work showed that /akta/, created by splicing together appropriate parts of /aka/ and /ata/, are perceived as /ata/ when the stop closure duration approximates that of a single medial stop; at long stop durations, both stops are heard. Apparently listeners need to integrate the multiple acoustic events into a single percept that makes sense for their language. At short closure duration, production constraints dictate that there is only time for one stop, hence the weaker cues for the first stop are disregarded. The present study reinforces this interpretation by showing that voiceless consonant-spliced clusters, e.g., /akta/, require a longer closure interval to be heard as two segments than do voiced ones, e.g., /agda/, because in natural speech voiceless stop closures are longer than voiced stop closures. Perception, then, is guided by listeners' expectations that are further based on naturally occurring speech events.

**F2. Effects of varying chord progressions on P3 event-related potentials in musicians and nonmusicians.** Sarah W. Chuang, Robert D. Frisina (Department of Physiology, University of Rochester, Rochester, NY 14642), Gary C. Crummer (Department of Otolaryngology, University of Rochester, Rochester, NY 14642), and Joseph P. Walton (Eastman School of Music, University of Rochester, Rochester, NY 14642)

Event-related potentials were recorded from the scalps of both musicians and nonmusicians while actively engaging in an auditory discrimination task. Using a classical "oddball" paradigm [Donchin, *Psychophysiology* **18**, 493 (1981)], subjects ( $N = 19$ ) listened to one series of pure tones and four series of musical chord progressions in which the second of four chords differed for the target stimuli. The level of discriminability for the progressions relied upon the degree of change of the target chord in terms of melodic interval, sonority, and voicing alteration in the alto and soprano lines. The P3 waveforms displayed greater latencies and widths for chord progressions relative to the pure tones. For the chord progressions, musicians had a higher percentage of hits compared to nonmusicians, with the former group showing a significant inverse correlation between number of hits and P3 latency. Additional observations were that the P1 potential for the first chord of the progressions was considerably greater in amplitude than the P1 for the three later chords. Also, P1 amplitudes were greater for musical sounds compared to pure tones, and the negative wave immediately following P1 showed a larger amplitude for the pure tones relative to the chords. The results suggest that melodic interval and voicing cues differentially affect neural potentials generated from central auditory and memory systems. Different cognitive strategies related to musical training may underlie these observations. [Work supported by Eastman School of Music, Program in Biology and Medicine, Audiology Division, University of Rochester.]

**F3. Vibrational modes of a baritone guitar.** Juan Lozano, Eric T. Watson, and Thomas D. Rossing (Northern Illinois University, DeKalb, IL 60115)

This work studied the principal modes of vibration of a baritone guitar with steel strings tuned to  $A_1$ ,  $D_2$ ,  $G_2$ ,  $C_3$ ,  $E_3$ , and  $A_3$ , a fifth below those of the usual folk guitar. Likewise, the corresponding resonances are designed to be a fifth below those of a steel-string folk guitar [R. E. Ross and T. D. Rossing, *J. Acoust. Soc. Am.* **65**, S72 (1979)]. The modal shapes have been determined by scanning the plates with an accelerometer and also by scanning the nearfield sound with a probe microphone. Sound radiation fields in an anechoic room for several of the modes are also shown.

**F4. An automated portable transducer test system.** Karl W. Rehn and Gregory L. Cannon (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin TX 78713-8029)

A set of computer programs was developed to control an automated portable transducer test system. The system hardware, consisting of commercial test equipment and a custom-designed amplifier/attenuator unit, was controlled by a Hewlett-Packard integral personal computer. Pulsed measurements were used for all tests to decrease the effects of self-resonances and reflections, which can corrupt continuous-wave (cw) measurements. This computer-automated transducer test system offers the user a wide selection of tests, such as immittance loops, impedance and phase plots, dc resistance (megohm) measurements, discrete frequency measurements, and FFT analysis. Besides being able to perform each type of test singly, the user can combine test types and setup parameters into "canned" routines in which the system executes a user-defined test sequence. The structure and content of the overall test system are discussed, and a qualitative discussion of the system's efficiency, accuracy, and flexibility is given. The limitations of the system and sample test configurations and output are also presented [Work supported by NAVSEA.]

**F5. Nonlinear sound scattering of crossed focused beams in the presence of turbulence.** Stephen C. Rife and Murray S. Korman (Department of Physics, United States Naval Academy, Annapolis, MD 21402)

Experiments are performed involving the interaction of mutually perpendicular crossed ultrasonic beams overlapping and interacting in the presence of turbulence in water. The turbulence is created by a  $d = 0.64$ -cm-diam submerged water jet with nozzle exit velocity 13.3 m/s. A profile of sum frequency pressure  $p_+$  versus distance, with characteristics of turbulent velocity, is determined by scanning across the width of the jet with the two cw acoustic beams (of frequencies  $f_1 = 1.9$  MHz,  $f_2 = 2.1$  MHz) focused on a point in the jet. The focal lengths are 15.2 cm for both sending units. In scanning, the receiving transducer unit (unfocused and located 15.2 cm from the interaction region) moves along with the senders so that propagation distances never change. The receiver axis is perpendicular to the jet. Scattered sum frequency pressure profile scans (at distances of  $16d$  and  $34d$  from the nozzle) compare well with known values of the radial turbulent rms velocity profile. Statistical properties of skewness  $s$  and kurtosis  $k$  from fluctuations in  $p_+$  are measured as a function of radial distance. [Work supported by the Naval Academy Research Council.]

**F6. Interaction of acoustic and thermally induced instabilities.** Stacey West and Charles Thompson (Department of Electrical Engineering, University of Lowell, One University Avenue, Lowell, MA 01854)

Thermal convection in a low Prandtl number fluid can exhibit both steady and unsteady flow states. The state to which the fluid is attracted is dependent on the ratio of the body forces to the viscous forces. This ratio is called the Rayleigh number. Above a critical value of the Rayleigh number it has been shown that unsteady thermal modes cut on and cut off in discrete ranges of temperature with increasing value of the Rayleigh number. In this paper, results of an experimental investigation into how high-frequency acoustics waves interact with these unsteady thermal modes are presented.

**F7. An algorithm for obtaining immittance or impedance loops.** Gregory L. Cannon and Karl W. Rehn (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

An immittance loop algorithm has been developed for automated sonar transducer testing. This algorithm produces an excellent loop given only two frequencies separated by a resonance. Measurements of impedance and phase are made at increasing frequencies until the difference between the last recorded phase value and the current phase value approaches a calculated phase increment. At this point the new data are recorded and a new phase increment is calculated. Measurements continue for increasing frequencies until the specified upper frequency is reached. When plotted as a loop, the recorded data can be used to determine resonance frequencies, bandwidth, and other properties of resonant circuits. The algorithm has been implemented on two computer-controlled measurement systems with excellent results. Benefits of this algorithm include measurement of devices that have multiple resonances, measurement of high-Q devices, wide frequency range of operation, very efficient use of data space, and ease of implementation. The algorithm and its limitations are discussed, and sample programs and output are presented. [Work supported by NAVSEA.]

**F8. Scattering of a focused sound beam by turbulence.** Mary M. Jackson and Murray S. Korman (Department of Physics, United States Naval Academy, Annapolis, MD 21402)

Experiments are performed involving the scattering of a cw beam of focused ultrasonic waves by turbulence in water. The turbulence is created by a  $d = 0.64$ -cm-diam submerged waterjet with a nozzle exit velocity of 13.3 m/s. A mechanical apparatus positions a focused beam of sound (of frequency 1.9 MHz) to scan across the width of the turbulent waterjet plume. The sending transducer unit has a focal length of 15.2 cm and is always directed at an angle of  $45^\circ$  to the jet axis and aimed "downstream."

A transducer receiving unit (fixed relative to the sender) is directed perpendicular to the jet axis. This unit is unfocused and is located 15.2 cm from the focal point. The scattering angle between sender and receiver is  $45^\circ$  in the "upstream" direction. Profiles of scattered pressure  $p_s$  as a function of the radial scanning distance  $r$  are compared to known turbulent rms velocity profiles at distances of  $16d$  and  $34d$  from the nozzle exit. Measurements of the spectral broadening and Doppler shift of the scattered pressure taken at points across the width of the jet will be used to predict the local values of the rms turbulent velocity and the mean jet velocity, respectively, across the width of the jet. [Work supported by the Naval Academy Research Council.]

**F9. Acoustic remote sensing of the upper ocean.** J. Shorey and N. P. Chotiros (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

There are particles suspended in the ocean, such as bubbles and biological debris, which can backscatter acoustic waves. It is postulated that the backscatter from a collection of particles produces a unique acoustic signature. Since these particles are passive riders, it would be feasible to track the movement of the water by using a sonar and a cross-correlation method. The volume scatterers and the operation of an acoustic tracker were simulated on a computer in FORTRAN. Then a signal simulator was used to generate a set of backscattered signals for a multibeam sonar system. The scatterers were ensonified as their positions were changed in a way that simulated flows and eddies in the water. The signals from successive pings were cross correlated, and the resulting peaks in the correlation matrices were used to estimate movement. A reasonably accurate portrayal of the water motion has been demonstrated.

**F10. Passive damping with differential density fluid damper.** J. Bart, C. Morris, and M. Gormley (The Catholic University of America, Mechanical Engineering Department, Washington, DC 20064)

This project presents a new class of passive fluid dampers that uses the differential density between floating spheres and a damping fluid to provide high damping characteristics over a wide frequency band. The developed damper is not only light in weight but also effective in damping multidirectional vibrations. These features make it superior to the fully filled or the partially filled fluid loop dampers that are routinely used to damp out the vibration of spinning spacecrafts and satellites. The effect of varying the design parameters of the damper on its dynamic characteristics is investigated in an attempt to determine the optimum values of these parameters. The considered design parameters are, namely, the viscosity of the damping fluid and the concentration of the floating spheres. The performance of the damper alone is determined when subjected to step and sinusoidal vibrations as well as when it is integrated in a large space structure.

## Session G. Psychological Acoustics I and Physiological Acoustics I: Experimental Techniques (Poster Session)

William M. Hartmann, Cochairman

*Department of Physics and Astronomy, Michigan State University, East Lansing, Michigan 48824-1116*

Robert L. Smith, Cochairman

*Institute of Sensory Research, Syracuse University, Syracuse, New York 13244-5290*

### Contributed Papers

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#### G1. Some research "toys." Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007)

An unbridled curiosity and a certain lack of self-discipline have resulted in a series of products whose research usefulness sometimes exceeds their immediate economic value to the author's company. On display will be low-noise, low-vibration-sensitivity microphones for measuring spontaneous and stimulated cochlear emissions; a  $\frac{1}{2}$ -in. microphone with switchable diffuse-field-inverse filter for simplified KEMAR™ measurements; insert earphones with 70–100 dB of interaural isolation, 30–50 dB of noise exclusion, and various frequency responses (flat pressure response at the eardrum to beyond 12 kHz, flat diffuse-field-referenced response, and TDH-39-like response); and a 1-mm-o.d. probe microphone with flat frequency response beyond 10 kHz and two types of precision tubing (soft silicone rubber with approximately 35-dB wall attenuation and semirigid polyethylene with approximately 50-dB wall attenuation).

#### G2. Hand-held auditory screener. Lynn S. Alvord (Department of Communication Disorders, University of Utah, 1201 Behavioral Science Building, Salt Lake City, UT 84112)

A new hand-held auditory screener is presented that utilizes filtered square waves as test signals. Circuit design has resulted in a total size of  $2\frac{1}{2} \times 4 \times 2\frac{1}{2}$  in., including earphone and cushion. High-frequency filtering of the square wave signal results in acceptably low levels of harmonic distortion according to ANSI specifications for audiometers [ANSI S3.6-1970]. Low current drain design provides for a "battery-low" and "signal-on" indicator. Individual intensity pots allow for periodic calibration or modification of presentation levels between 20 and 40 dB HL. Initial clinical trials with 50 patients show good agreement with conventional, pure-tone audiometric screening. Rationale for increased use of hearing screening by health care practitioners is presented.

#### G3. Instrumentation for dolphin echolocation experiments. Whitlow W. L. Au (Naval Ocean Systems Center, P.O. Box 997, Kailua, HI 96734)

The use of personal computers has contributed significantly in dolphin echolocation research by providing an inexpensive instrument to measure echolocation signals, monitor and control experimental devices, and to store data. Dolphins typically emit short duration (50–100  $\mu$ s), broadband click signals with repetition rates that can vary from tens of clicks per second to several hundred clicks per second. In this paper, three electronic measurement devices used with Apple II computers in dolphin echolocation experiments will be discussed. The first device measures the number of clicks emitted, the intervals between clicks, the peak-to-peak amplitude of each click, and the time of activation of various switches. The second device measures the frequency spectrum (between 30 and 135 kHz) of each emitted click signal in real time, with a resolution of 15 kHz.

The third device is a phantom electronic target simulator that digitizes each emitted click and then retransmits the click signal under control of the computer. The development of these three devices has been a process of evolution in sophistication over a period of several years of dolphin echolocation research conducted at the Naval Ocean Systems Center.

#### G4. High-frequency intensity-calibrated threshold measurements using a two-microphone system. William J. Murphy, Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47904), and Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

As a means of calibrating high-frequency hearing tests without the problems inherent in the use of pressure-calibrated headphone systems [M. R. Stinson, *J. Acoust. Soc. Am. Suppl.* 1 87, S75 (1987)], an intensity-calibrated method of measuring hearing thresholds based on the two-microphone technique has been developed [W. J. Murphy, A. Tubis, and G. R. Long, *J. Acoust. Soc. Am. Suppl.* 1 87, S75 (1987)]. The two-microphone probe is used to estimate the acoustic impedance and the power reflection coefficient of the middle and inner ear system as well as to calibrate psychoacoustic measurements. Various checks on the reliability of the long-term phase calibration are discussed. A computer simulation of the ear canal acoustics is used to determine an appropriate downstream distance of the probe tips from the insert-earmold sound port. Typical data are given for both objective and psychoacoustic measures obtained with the two-microphone system. [Work supported by a grant from the Deafness Research Foundation.]

#### G5. Modified DigiSound-16 becomes a precision digital oscillator. W. M. Hartmann, D. L. Edmunds, T. V. Atkinson, (Physics Department, Michigan State University, East Lansing, MI 48824), and Hal Chamberlin<sup>1</sup> (Microtechnology Unlimited, 156 Wind Chime Court, Raleigh, NC 27615)

The DigiSound-16 is a two-channel, 16-bit DAC-ADC system with a 32K word buffer. Because of its high performance and low cost it is an attractive signal source for psychoacoustic experiments. The device can be modified to serve as a precision digital oscillator by using the technique of fractional addressing [W. M. Hartmann, *J. Acoust. Soc. Am.* 82, 1883–1891 (1987)]. The frequency resolution is 0.003 Hz over the audible range and the total distortion approaches the theoretical limit of  $-85$  dB for single-channel operation. The modification requires that the 15-bit address register be replaced by a 24-bit increment-and-add circuit. Two-way communication between the modified device and the host computer allows the address increment to be changed continuously so that the oscillator can be manually tuned, as in the method of adjustment. The portion of the buffer that is recycled can be reduced by factors of 2, under program control, to as little as 2K words. The reduced buffer length promotes

efficiency while compromising frequency resolution and distortion figures. [Work supported by the National Institutes of Health.] <sup>\*)</sup> Present address: Kurzweil Music, 411 Waveley Oaks Rd., Waltham, MA 02154.

**G6. Using MIDI, the standard Musical Instrument Digital Interface, for psychoacoustic experiments.** H. E. F. Williams, G. L. Gibian, E. N. Hamden, and A. Evans (Physics Department, The American University, Washington DC 20016)

MIDI, the music industry's standard Musical Instrument Digital Interface, allows control of commercially available music synthesizers by personal computers, as well as measurement and reproduction of musical performance parameters. The resolution in frequency, amplitude, timbre, and time of the system is adequate for various psychoacoustic experiments. A Yamaha DX7 synthesizer controlled by an IBM PC was used to measure subjects' judgments of the similarity of synthesized tones differing in timbre [G. L. Gibian *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **82**, S68 (1987); G. L. Gibian *et al.*, *Audio Engineering Society Preprint #2488* (1987)].

**G7. Demonstration of auditory impedance/reflectance measurement technique, and MIDI enters the research lab.** Douglas H. Keefe (Systematic Musicology Program, School of Music, DN-10, University of Washington, Seattle, WA 98195)

A microcomputer-based implementation of a two-microphone wave-tube technique for measuring acoustical impedance and energy reflectance will be demonstrated. Data can be fully analyzed with graphical output while the subject is present, thus giving the researcher important flexibility. The Musical Instrument Digital Interface (MIDI) is a standard in electronic music with potential research applications in the areas of acoustical stimulus generation, instrumentation control, and data collection. Selected applications will be demonstrated on the microcomputer with MIDI peripheral devices, and a MIDI software library designed for integration into the research setting will be described. [Work partially supported by NINCCDS.]

**G8. A programmable subject response box for psychophysical experimentation.** Robert Ling and Edward M. Burns (Department of Speech and Hearing Sciences, University of Washington JG-15, Seattle, WA 98195)

Software that utilizes a commercially available touch pad ("Koala Pad") as an all-purpose subject response box will be demonstrated. [Work supported by NINCCDS.]

**G9. A computer interface for psychophysical and speech research with the Nucleus cochlear implant.** Robert V. Shannon, Doug D. Adams, Roger L. Ferrel, Robert L. Palumbo, and Michael Grandgenett (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131)

A custom interface was designed and implemented that, under program control, allows presentation of any available pulse stimulus to a patient fitted with the Nucleus implant. The interface connects to a standard parallel output port of a PC or AT compatible computer. The host computer sends a stream of bytes to the parallel port that specify the configuration of the desired output pulses. Upon receipt of the data, the interface generates the appropriate pulse-coded sequence to deliver the specified pulses to the patient's external coil. This interface makes it possible to interleave pulses on two or more electrodes, to modulate the amplitude or timing of a pulse sequence, or to sweep a stimulus across the electrode array. This interface allows investigators to conduct psychophysical and speech experiments that cannot be performed with the standard speech processor interface (SPI) normally used to set the patient's device. [Work supported by NIH.]

**G10. IDENTIFY: A program for speech recognition, identification, and categorization experiments on PC compatible computers.** Robert V. Shannon, Robert L. Palumbo, and Michael Grandgenett (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131)

IDENTIFY is a program written in Turbo Pascal that accepts any combination of up to 100 speech waveform files as stimuli and up to 25 response categories. The waveform files can be from either ILS or C-Speech, or any compatible format. The program presents the stimuli in block-randomized order and waits for the subject to respond on the touch sensitive surface of a small digitizing tablet (Koala Pad). The tablet can be configured to reflect any number and pattern of desired responses. The most common case is when the number of stimuli and responses are the same, producing a standard confusion matrix. In this special case the program outputs an additional data file that conforms to the format for the Speech Information Transmission Analysis (SINFA) [Wang, *Behav. Res. Methods Instrum.* **8**, 471-472 (1976)]. A categorization experiment can be run by specifying many stimuli that span a continuum, allowing only two responses. The program requires a PC or AT compatible computer, a Koala Pad connected to a game port, and a Data Translation DT2801-A D/A output board. [Work supported by NIH.]

**G11. Transducer and transducer measurement system for the PC.** J. B. Allen (Acoustics Research Department, AT&T Bell Laboratories, Room 2D530, Murray Hill, NJ 07974)

The measurement of several acoustic transducers using a measurement system that is implemented on the PC will be shown. This system uses the ARIEL DSP-16 and a PC-6300. The software developed for this purpose, which is called SYSid, was developed first about 8 years ago, and in the past few years was ported to the PC. SYSid will measure the frequency response of a transducer in less than 100 ms with an accuracy that is determined by either the nonlinearities or the noise in the system. Typically, this noise and distortion is more than 80 dB down, giving results that are accurate to 1 part in 10<sup>4</sup>. SYSid also will measure the phase, group delay, impulse response, and the distortion as a function of frequency and level. In the demonstration, an attempt will be made to measure the acoustic impedance of a section of tube, and a rho-c screen.

**G12. Software for analyzing and plotting scientific data.** Walt Jesteadt, Stephen T. Neely, Brian P. Callaghan, and Roger L. Ferrel (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131)

Our PLT program, in use in a number of laboratories, translates a high-level data plotting command language into Tektronix or HP-compatible output. It is specifically designed for scientific applications, allowing maximum flexibility in the use of partial log axes, labels, and error bars, with rapid output on a graphics terminal or publication quality hardcopy on a pen plotter or laser printer. GREG, a spreadsheet data analysis program developed in parallel with PLT, allows data to be represented in terms of a five-dimensional factorial design. It provides convenient data entry from either keyboard or digitizer pad, automatic handling of missing values, data transformations, low-level statistical analyses, and flexible generation of tables and plots. Plots are generated by producing PLT input files. GREG can call user supplied programs to fit specific models or perform other functions, so the data analysis framework can be easily expanded. Both PLT and GREG run under RT-11, TSX, UNIX, and MS-DOS; they will be demonstrated on a portable AT clone. [Work supported by NIH.]

**G13. Application of digital signal processing techniques to research on sound localization.** Frederic Wightman and Doris Kistler (Department of Psychology and Waisman Center, University of Wisconsin, Madison, WI 53705)

The availability of fast, general-purpose laboratory computers has dramatically changed the way research is done in many areas of psychoacoustics. In studies of sound localization, for example, standard digital signal-processing techniques are used to synthesize stimuli for headphone presentation that contain all of the localization cues (e.g., interaural differences, pinna effects) that are available from sounds presented in free field. This brings to localization research a much needed degree of stimulus control and specificity, and allows us to address certain basic issues that were heretofore inaccessible. This presentation will survey the general principles involved in the application of digital signal processing to localization research, and will discuss the practical limitations of the specific techniques that are used (e.g., FFTs, FIR filters, inverse filtering). Empirical data will be shown in order to address questions relating to the

classic tradeoffs among signal/noise ratio, bandwidth, word-length, and sampling frequency. The feasibility of real-time digital processing in currently available PC-based systems will also be discussed. [Work supported by NIH, NSF, and NASA.]

**G14. Classical respiratory conditioning used in auditory psychophysics of the goldfish: Methods and results in detection and discrimination studies.** R. Fay, S. Coombs, and C. Wheelers (Parmlly Hearing Institute, Loyola University of Chicago, 6525 N. Sheridan Road, Chicago, IL 60626)

Classical conditioning of respiration has been used to study many aspects of hearing in the goldfish, including absolute and masked thresholds, frequency, intensity, and time discrimination thresholds. A restrained fish is presented with a 7-s auditory signal that ends with a brief electric shock across the body. Shock causes an unconditioned suppression of respiration lasting several seconds. Several pairings of the auditory signal with the shock results in respiratory suppression during the signal. Initial conditioning is rapid (conditioned responses appear within the first 10 to 20 trials), several thresholds can be obtained in one day, and thresholds can be obtained for individual animals for several years. Critical factors for success with this method include the measurement of respiration (using a thermistor), method of animal restraint, levels of electric shock, intertrial intervals, false alarm estimation, overall respiratory rate, subject selection, and water conditioning. Details of the methods and procedures that have been found useful in conditioning and threshold definition will be given along with illustrative data on threshold values and stability in masking and intensity discrimination experiments.

**G15. Psychoacoustical measurements with delays in neonates' vocalizations.** Lincoln Gray (Department of Otolaryngology, University of Texas Medical School, Houston, TX 77030)

Newborn chickens momentarily delay their usually incessant peeping when they hear either an onset or change in an auditory stimulus. This unconditioned response provides several ways to study the early development of hearing. (1) Receiver operating characteristics, drawn from pooled histograms of responses on stimulus and control trials, strongly suggest that this response is a measure of auditory detection. Areas under these curves can be used to quantify the auditory abilities of newborn subjects. (2) Adaptive procedures based on paired comparisons of stimulus and control trials provide rapid estimates of thresholds and difference limens. A five-frequency audiogram, for example, can be obtained from a newborn chick in several minutes. (3) Responses to all possible pairs of transitions between multiple stimuli can be used as proximity data for multidimensional scaling algorithms. This provides a "map" of how auditory perceptions change immediately after birth. (4) Neonates' responses to naturalistic stimuli may be different than those to more arbitrary pure-tone and noise signals. Delayed vocalizations also occur in other animals, allowing comparative studies of auditory development. [Work supported by NIH.]

**G16. Techniques for free-field testing of spatial attributes of acoustic signals.** R. Wayne Gatehouse (Department of Psychology, University of Guelph, Guelph, Ontario N1G 2W1, Canada)

For a number of years Gatehouse and his coauthors have reported on studies of various aspects of spatial acoustics (localization, depth perception, masking) done under free-field conditions of signal presentations. Basically, the paradigms have involved comparisons of the responses of normally hearing, artificially degraded (monaural or binaural occlusion by ear muffs and plugs), and real hearing impaired subjects of various types and degrees of loss, under conditions that presumably mirror the more normal hearing environment, i.e., reverberant or semireverberant rooms, all types of signals (noises, tones, speech), from positions that vary in both azimuth and elevation and from all around the subjects. In this session, some of these techniques and conditions will be displayed in a more detailed manner than is usually available in verbal presentations of the methodologies. [Work has been supported under various NSERC and MRC grants.]

**G17. The use of multidimensional scaling techniques for revealing perceptual categories for complex stimuli in animals.** Robert J. Dooling, Kazuo Okanoya, Susan D. Brown, and Thomas J. Park (Department of Psychology, University of Maryland, College Park, MD 20742)

A combination of operant conditioning and multidimensional scaling techniques for demonstrating natural perceptual categories for complex sounds in small birds are described. Birds are trained using operant conditioning procedures on either a same/different discrimination task or on a task requiring the detection of change in a repeating background. Response latencies are used to construct similarity matrices, and multidimensional scaling procedures are then used to produce spatial maps of complex sounds reflecting perceptual organization. Stimulus similarity is represented by spatial proximity. Stimulus groupings in multidimensional space indicate perceptual categories that can be confirmed by cluster analyses. These procedures have been used to study the perception of complex sounds such as bird calls and speech in small birds, but these techniques should also be useful in examining the perception of complex sounds in other animals. [Work supported by NIH.]

**G18. Small-sample statistical analysis of Levitt's psychophysical procedure.** Brent W. Edwards and Gregory H. Wakefield (Department of Electrical Engineering and Computer Science, University of Michigan, Ann Arbor, MI 48109-1109)

Levitt's 2IFC adaptive procedure is a widely used psychophysical method for estimating detection or discrimination threshold. The commonly accepted analysis shows that the average of the levels at which reversals occur approaches the 0.707 point on the psychometric function when the levels are updated according to the "2-up, 1-down" rule. This analysis is based on a continuous representation of the psychometric function and on the asymptotic behavior of the tracks. The more practical case was considered in which the psychometric function is represented by a set of discrete levels and in which the stopping criterion is a fixed number of reversals. A Markov model for the reversal levels is proposed as a method for determining the statistics of the estimator. Based on this approach, the estimator is significantly biased for small numbers of reversals, and decreases monotonically as this number becomes large. Convergence to the 0.707 point, however, is not guaranteed and depends on the sampling of the psychometric function. These results hold, in general, regardless of the form of the psychometric function or the method by which the psychometric function is sampled. More rapid convergence with less bias may be achieved by tailoring the sampling to the psychometric function. Several approaches to the design of such sampling will be discussed. [Research supported by AFOSR.]

**G19. Positron emission tomography (PET) as a technique for studying patterns of regional cerebral function associated with auditory perception and speech production.** John J. Sidtis, Vijay Dhawan, James R. Moeller, Stephen C. Strother, David Eidelberg, and David A. Rottenberg (Department of Neurology, Memorial Sloan-Kettering Cancer Center, New York, NY 10021)

PET is an *in vivo* technique for the measurement of patterns of regional cerebral blood flow (rCBF), metabolic rate for glucose (rCMRGlu), and the uptake of neurotransmitters such as dopamine. Scanning times range from 1 min for rCBF, to 2 h for dopamine, with rCMRGlu scans requiring 1 h. Studies of rCMRGlu in patients with movement disorders and dysarthria have demonstrated correlations between voice onset time abnormalities and rCMRGlu in the basal ganglia; similar studies with dopamine are currently in progress. Because of the shorter scan time, rCBF studies are better suited for perceptual studies. Using a steady-state C0150 inhalation technique, 14 1-min scans can be obtained back-to-back. The C0150/PET has been used to study rCBF patterns in within-subject studies that include both stimulated and unstimulated periods, with stimuli such as broadband white noise, music, and complex tones in a pitch discrimination task. Results to date demonstrate that both regional and global changes are associated with stimulation states. PET provides a unique technique for studying human cerebral physiology associated with auditory perception and speech production. [Work supported by NIH.]

**G20. Acetylcholine effects on tone elicited single unit responses in rat auditory cortex.** H. K. Rucker (Department of Physiology, Meharry Medical College, Nashville, TN 37208)

Urethane (1.5 g/kg) anesthetized male Sprague-Dawley albino rats were affixed with a head bolt and a burr hole placed over the temporal cortex and wound edges infiltrated with lidocaine before being placed in an electrically shielded, sound attenuating chamber. Tone bursts were generated with a software controlled M308 stimulus generator and 300 stimulus control module (Modular Instruments, Inc.) and sent via a secondary amplifier to a Realistic 40-1377 loudspeaker. Sound-pressure levels were determined with a calibrated  $\frac{1}{4}$ -in. B & K 4135 condenser microphone. Single unit activity was recorded via the center barrel (4M NaCl) of glass multi-pipette. Drug barrels contained acetylcholine chloride (0.5M, pH 4.5), atropine sulfate (0.02M, pH 5.0), and glutamic acid (monosodium salt, 0.2M, pH 8.0). Sixty-six percent ( $n = 21$ ) of units characterized demonstrated an atropine sensitive Ach facilitated response to near best frequency tone bursts. An atropine insensitive Ach induced response decline was noted in 13% ( $n = 4$ ). [Work supported by NSF Grant BNS 8617937.]

**G21. A new set of auditory demonstrations.** A. J. M. Houtsma (Institute for Perception Research, P. O. Box 513, 5600 MB, Eindhoven, The Netherlands), T. D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115), and W. M. Wagenaars (Institute for Perception Research, P. O. Box 513, 5600 MB, Eindhoven, The Netherlands)

A new series of auditory demonstrations for classroom use, modeled after the well-known "Harvard Tapes" [D. M. Green, Harvard Laboratory of Psychophysics (1978)] was developed and produced. The demonstrations are issued on a compact disc (CD), are accompanied by a 92-page explanatory booklet, and fit conveniently in a handy slide box. Almost all demonstrations were synthesized digitally on DEC Vax 11/780 and MicroVax II computers, converted to analog signals with DSC-200 16-bit two-channel D/A converters, and recorded on 16-bit PCM video tape. Brief spoken introductions to each demonstration were recorded on similar PCM video tape. Editing of the master tape was done digitally in 1630 format on  $\frac{1}{4}$ -in. U-Matic tape, from which the CD master was made. The CD medium, allowing a S/N ratio of up to 90 dB, is very appropriate for sound demonstrations that require a clean acoustic background. The medium also provides random access to individual demonstrations or parts thereof, allows preprogramming of any desired combination of demonstrations, and is much more resistant against wear and tear than traditional tapes or records. The CD can be played on any CD player capable of handling selections from up to 80 tracks. The demo set is available through the ASA.

**G22. An auditory filterbank design and its hardware implementation with DSP.** Takashi Komakine and Tatsuya Hirahara (ATR Auditory and Visual Perception Research Laboratories, Twin 21 Building, MID Tower 2-1-61 Shiromi, Higashi-ku, Osaka 540, Japan)

As the first step in constructing a signal processor to simulate the human auditory function, a computational filterbank with cochlear frequency analysis function was designed [T. Komakine and T. Hirahara, Tech. Rep. IEICE Jpn. SP87-45, 65-72 (1987) (in Japanese)] and its hardware with DSP implementation completed. From the various cochlear filtering models proposed to date, a cascade/parallel type filterbank model [e.g., R. F. Lyon, ICASSP82, 1282-1285 (1982)] was chosen followed by design of a filterbank composed of 61-channel IIR filter blocks. Each filter block was spaced at  $\frac{1}{3}$  Bark intervals from 1 to 21 Bark. Param-

eters for each filter were decided by taking into account observed physiological data and filter stability. The parameter optimizing effects of the filter were discussed. Sound spectrograms produced by using the filterbank showed that this auditory filterbank model allows excellent frequency and time resolution. A hardware implementation of the filterbank, composed of floating-point DSP chips and a host controller, was studied. In addition to the computational filtering functions, some control functions, such as automatic addressing previously done in the host controller, are distributed to each DSP. Therefore, control of each DSP unit by the host is easier. More sophisticated functions are easily added. This is in part due to a program memory commonly accessible both from each DSP as well as the host. Finally, the auditory filterbank is capable of a frequency analysis of up to 3000 channels within 5 ms, which is equal to the time required for the wave to travel the basilar membrane. It is also expandable to allow hardware implementation with more than 30 000 cascaded DSP units.

**G23. Whistling with heavy gases or at elevated pressures.** R. Stuart Mackay (San Francisco State University, San Francisco, CA 94132)

Most divers can speak easily but cannot whistle at elevated pressure. Suggestively, sperm whales are not known to whistle. However, rigid flutes or bird calls remain easy to blow "at depth"; frequency changes little. The author can just whistle tunes at 4 atm (equivalent water depth of 30 m). At one atmosphere he could whistle with a breath of half air and half sulfur hexafluoride but not with pure SF<sub>6</sub> (density relative to air of 5.1); alternate whistling in and out aids the observation. Possibly this is due to increasing gas density yielding damping vibrations in the soft tissues shedding vortices. Caution: Limiting duration prevents suffocation; possible impurities in SF<sub>6</sub> are toxic and corrode metal. The SF<sub>6</sub> lowers whisper and speech tones but not hum and whistle frequencies; reflex changes in the throat can confuse the latter. A colleague, Jon Pegg, can whistle tunes normally as well as through his nose with mouth closed and vocal cords constricted to a slit. His nose whistle quenches at 20 m and the normal one at 10 m, suggesting sites of different stiffness. It is said to be impossible for organ pipes to sound if the stream is denser than the surroundings. Several blew well (frequency lowered) with octafluoropropane (density 6.5) in air.

**G24. Digital synthesis of binaural auditory localization azimuth cues using headphones.** Richard L. McKinley and Mark A. Ericson (Armstrong Aerospace Medical Research Laboratory, Wright-Patterson AFB, OH 45433)

A laboratory demonstration prototype of a digital auditory localization cue synthesizer has been developed. This synthesizer uses a single audio input that is separately processed in real-time for independent presentation to each ear using headphones. The headphone presented acoustic signals are easy to localize and appear to be out of head. The acoustic image is stabilized for head movement by use of a three-space head tracking device. The paper will describe the salient parameters of the design. A description of the psychoacoustic and electroacoustic measurements that led to the design will be presented. Human performance data on free field, simulated, and synthesized localization cues will be described and a real-time interactive demonstration will be available for interested listeners.

## Session H. Engineering Acoustics I: Transducers and Their Applications

George S. K. Wong, Cochairman

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

Caroline Fu, Cochairman

Boeing Aerospace, P. O. Box 3999, MS:82-58, Seattle, Washington 98124

### Contributed Papers

9:15

**H1. Fiber optic flexural disk microphone.** T. Hofer, D. A. Brown, and S. L. Garrett (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

An interferometric fiber optic microphone consisting of a 10-m-long, 4-cm-diam, flat-wound (spiral) single mode optical fiber bonded to a simply supported, 8-cm-diam, 3-mm-thick aluminum disk will be described. In the presence of a pressure difference across the plate or an acceleration of the plate, the surface strain induced in the plate is transmitted to the optical fiber. The optical phase shift induced by the strain is detected in an all-fiber Michelson interferometer. The calculated strain [A. E. H. Love, *A Treatise on the Mathematical Theory of Elasticity* (Dover, New York, 1944), 4th ed., Sec. #309] and optical strain measured using static pressure, acoustic pressure, and a calibrated accelerometer, are in good agreement ( $\pm 10\%$ ) and yield a sensitivity of 30 milliradians per Pascal per meter of optical fiber below the plate resonance frequency of 3 kHz. [Work supported by the Office of Naval Research and Office of Naval Technology.]

9:30

**H2. Anomalous degradation of electromechanical coupling and resonance frequency in tonpizl transducers.** Michael P. Johnson (Acoustics and Sensor Systems Group, Systems Engineering Department, Gould, Inc., Ocean Systems Division, 18901 Euclid Avenue, Cleveland, OH 44117)

In recent years, the bandwidth of sonar arrays has been increasing, requiring wide bandwidth transducer element designs. Tonpizl transducers have come to be the dominant element design in such arrays. The location of the fronting blocks flexural resonance relative to the operating band becomes a critical feature of these wideband element designs. In some cases, unexpectedly low electromechanical coupling and resonance frequencies have occurred. These results are typically attributed to "cross modes" due to ceramic stack geometry or interface compliance at the stack ends, though no conclusive evidence exists for either. In a series of modeling investigations utilizing the piezoelectric capabilities of the commercial finite element code ANSYS, mode shapes at the fundamental resonance of a number of designs were compared. In certain designs, excessive and unsuspected bending motion in the fronting block was found to degrade the electromechanical coupling and resonant frequency. The results of these investigations will be presented along with some possible solutions.

9:45

**H3. Reciprocity calibration of an underwater transducer by the Delta-Z method.** R. Bedard<sup>1</sup> and S. R. Baker (Department of Physics, Code 61Ba, Naval Postgraduate School, Monterey, CA 93943)

A method for determining the free-field open-circuit voltage sensitivity of a reversible underwater electroacoustic transducer from the difference in its input electrical impedance when loaded by water and air was investigated theoretically and experimentally. An equation for the sensitivity was derived using reciprocal two-port network theory. The theory takes into account the diffraction due to the finite size of the transducer, its finite mechanical impedance, and its free-field radiation impedance in water. An experiment to test the predictions of the theory was performed using a 6-in.-diam, hollow, piezoelectric ceramic spherical transducer. The results of the experiment agreed within several dB with the results of a standard comparison calibration over the frequency range for which both the electrical impedance and comparison calibration data are considered reliable. The calibration method described, which has been termed the Delta-Z method, may be useful for *in situ* monitoring of transducer sensitivity in installations which can be flooded and purged. <sup>1</sup>Major, Canadian Armed Forces. Permanent address: 415 Squadron, C. F. B. Greenwood, Nova Scotia B0P 1N0, Canada. •

10:00

**H4. A four-sided flextensional transducer.** John L. Butler (Image Acoustics, Inc., P. O. Box 6, North Marshfield, MA 02059) and Kenneth D. Rolt (Raytheon Company, Submarine Signal Division, 1847 West Main Road, Portsmouth, RI 02871-1087)

A novel four-sided flextensional transducer has been constructed and tested for low-frequency operation. The transducer is in the shape of an asteriod formed by four curved concave metal plates driven at their junctions by four piezoelectric ceramic stacks configured as the spokes of a wheel. As the stacks expand in the positive cycle of operation, the four curved plates of the asteriod move outward in a motion that is the sum of the radial motion and the outward bending of the curved plates yielding a cumulative acoustic output. The experimental model tested is approximately 25 in. (0.635 m) in diameter and 6 in. (0.152 m) high with four curved steel plates, 0.25 in. (0.00635 m) thick, and operates in the frequency band from 500–1500 Hz. The results from the measurements show that current theoretical models for the conventional Class IV flextensional transducer may be used to estimate the performance of this transducer.

10:15

**H5. A magnetohydrodynamically driven Helmholtz resonant projector.** C. E. D. Haney, J. T. Newmaster, and S. L. Garrett (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

Unlike most conventional transducers that are noncompliant, a magnetohydrodynamic (MHD) transducer is ultracompliant since the volumetric velocity it generates is entirely determined by the acoustical impedance of the load it is driving [G. W. Swift and S. L. Garrett, "Resonance reciprocity calibration of an ultracompliant transducer," *J. Acoust. Soc.*



Am. 81, 1619 (1987)]. To exploit this property, a magneto-hydrodynamically driven Helmholtz resonator has been designed and calibrated that consists of a 400-ml-PVC tube with a 6-cm-long Lucite neck of rectangular cross section that is placed in the 1.0-cm gap of a 0.25-Tesla permanent magnet and filled with salt water. The transducer was calibrated using a

piezoelectric hydrophone within the resonator volume and by direct observation of the motion of the fluid in the neck using an MTI 1000-Fotonic (fiber optic) sensor. Operating at a resonance frequency of about 100 Hz with  $Q = 16$ , the transducer has a source strength of 5 cc/Amp-s. [Work supported by the Office of Naval Research.]

10:30-10:45

Break

10:45

**H6. Transfer functions of piezoelectric transducers for underwater applications and their inverse problems.** Shuchu Ren (Institute of Acoustics, Academia Sinica, Beijing, People's Republic of China)

Three related problems of the transfer functions of piezoelectric transducers are solved: (1) the theoretical calculation of the transfer function, (2) the reconstruction of the phase spectrum of the transfer function from the amplitude spectrum, and (3) the extrapolation of the transfer function to the full frequency range. The transfer functions of cylindrical and spherical piezoelectric transducers are obtained by solving the scattering problem of the transducer as a sound receiver. The calculations show that these transducers are typical minimum phase systems; consequently, the phase spectra of their transfer functions can be reconstructed from the amplitude spectra using the Hilbert transform. For an arbitrary piezoelectric transducer it is impossible to determine whether it is a minimum phase system. To solve this problem, an entirely different technique to reconstruct the phase spectrum is also developed. The phase spectrum of the transfer function of a F42-c spherical piezoelectric transducer is reconstructed from the measured values of its amplitude spectrum. The results are consistent with the measured values of its amplitude spectrum. The extrapolation of the transfer function of a piezoelectric transducer from the finite frequency range to the full range is done using the representation theorem of  $H^p$  functions. For a spherical piezoelectric transducer, the results obtained by extrapolation are in good agreement with the theoretical ones.

11:00

**H7. Characterization of length expander magnetostrictive rare earth-iron rods under normal use conditions in transducers.** F. Claeysen, D. Boucher, and S. Faure (GERDSM, DCAN Toulon, DCN, Le Brusac, 83 140 Six-Fours-Les-Plages, France)

Magnetostrictive rare earth-iron rods are presently mainly used as active elements of emission transducers. In this case, rods are mechanically prestressed and magnetically polarized. They are excited at high level around the transducer resonance frequency, which is much lower than the rod natural frequency. In order to characterize the rods under these conditions, a two-rod device has been designed, on which the electrical impedance has been measured before being processed by a computer program. The results are the longitudinal piezomagnetic constants of the alloy. The program is based on a plane-wave model; its assumptions on eddy currents and leakage flux are discussed with respect to results provided by a finite element computation. Coupling coefficient  $k_{33}$ , Young's modulus, permeability, and loss terms have been reckoned using this method for alloys of various origins and are presented. The characterization of materials started with L. Néel's laboratory alloy [J. Acoust. Soc. Am. Suppl. 1 81, S89 (1987)] and is carried on with grain-oriented alloys.

11:15

**H8. Acoustic interaction effects in unbaffled transducer arrays.** Christian Audoly (Groupe d'Etudes et de Recherche de Detection Sous-Marine Le Brusac, 83140 Six-Fours, France)

To calculate the transmitting characteristics of a transducer array, it is often necessary to take into account the acoustic interaction effects. For an array of pistonlike transducers set on planar surfaces, the model most commonly used is to assume the pistons set in an infinite rigid plane. This is not adequate when the array is not baffled, and when its dimensions are comparable to the wavelength. A theoretical model, using the combined Helmholtz integral equation method, is developed to calculate the mutual impedance matrix, the transmitting characteristics, and farfield directivities for arbitrary antenna geometries. Some results are compared to analytical models and to experiments. It is shown that the unbaffled condition produces important effects on the directivity functions and that the acoustic interactions modify the pressure distribution on the surface of the array.

11:30

**H9. Composite structure vibration analysis using the finite element method.** B. Hamonic, J. C. Debus, H. Tourneur (Institut Supérieur d'Electronique du Nord, 41 boulevard Vauban, 59046 Lille Cedex, France), and D. Boucher (Groupe d'Etude et de Recherche de Detection Sous-Marine, Le Brusac, 83140 Six Fours les Plages, France)

The need of low-frequency sonar transducers led to the concept of flextensional structure and, currently, promotes the use of composite materials. To design these transducers, a finite element modeling is very useful, because this method can accurately handle the various constituting parts as well as the radiation damping. The finite element code ATILA [J. N. Decarpigny *et al.*, J. Acoust. Soc. Am. 78, 1499 (1985)] has been used to analyze various types of metallic shell flextensional transducers and, more recently, a special three-dimensional composite finite element has been developed to take composite shells into account. This element is composed of unidirectionally reinforced lamina and various fiber orientations and materials can be used in the same element. The stiffness finite element matrix is calculated using equivalent elastic constants [S. W. Tsai, "Composite Design," in *Think Composites* (Dayton, Ohio, 1985)] and a separate numerical integration in three directions for each lamina. This paper presents the element formulation and its tests. Comparison between numerical and experimental results for simple structures has allowed the determination of the physical model parameters. Then, the modeling of a flextensional shell has demonstrated the accuracy of this approach.

**Session I: Bioresponse to Vibration I and Noise II: Combined Effects of Noise and Vibration on Humans**

Olavi Manninen, Chairman

*Department of Public Health, Medical Faculty, University of Tampere, P. O. Box 607, SF-33101 Tampere, Finland*

Chairman's Introduction—1:00

*Invited Papers*

1:05

**11. Complex environmental exposures and hearing functions.** Olavi Manninen (Department of Public Health, Medical Faculty, University of Tampere, P. O. Box 607, SF-33101 Tampere, Finland)

This study deals with changes in TTS<sub>2</sub> values, cardiovascular functions, haemodynamic activity, upright body sway, ratios of urinary catecholamines, and correlations between these changes in complex exposure situations. The study was based on a factorial experimental design with a total of 12 exposure combinations. Each individual experiment took 6 h with a pause of 1 h at noon. The subjects ( $n = 60$ ) were exposed to noise and whole-body vibration at two different dry bulb temperatures. The changes were dependent on the combinations of noise, vibration, and temperature to which the subjects were exposed. The TTS<sub>2</sub> values at 4 kHz were associated with the haemodynamic index (HDI) values when the subjects were exposed simultaneously to noise and stochastic vibration at 35 °C. The TTS<sub>2</sub> values at 6 kHz were associated most strongly with the HDI values after exposure to a combination of noise, stochastic, or sinusoidal vibration, and a temperature of 20 °C. The TTS<sub>2</sub> values at 4 and 6 kHz correlated positively with the noradrenaline/adrenaline (NA/A) ratio when subjects were exposed to noise at 35 °C. The association between the TTS<sub>2</sub> values and the 10A/NA ratio and especially the A/NA ratio was very strong when subjects had been simultaneously exposed to noise and sinusoidal or stochastic vibration at 35 °C. Furthermore, the highest positive correlation coefficients were found between the TTS<sub>2</sub> values at 4 kHz and the upright body sway values in the *X* direction when the subjects had been exposed to noise and sinusoidal vibration at 20 °C.

1:35

**12. The effects of noise and vibration on a complex task.** J. Sandover and C. S. Porter (Department of Human Sciences, University of Technology, Loughborough LE11 3TU, England)

Investigations of the interactive effects of noise and whole-body vibration on task performance have been dogged by such problems as the direct effects of vibration on control and the resistance of simple tasks to the environmental stress. The paper describes an attempt to overcome some of these problems and at the same time to consider conditions of practical relevance. Subjects were exposed to heat, noise, and vibration singly and in combination. The environmental magnitudes were typical of some occupations and exposure duration was approximately 6 h. Subjects were asked to perform a visual vigilance and decision making task, a visual monitoring and decoding task, and a battery of simple tasks. The former tasks were designed to place a significant cognitive load on the subject and strategy was emphasized by performance of both tasks at the same time. The tasks had relevance to a real work situation that the subjects were used to. Subjective and physiological measurements were also taken. The paper will present the results of the series of experiments just completed.

2:05

**13. The influence of whole-body vibration on noise-induced hearing loss: A review of animal experiments.** Roger P. Hamernik, William A. Ahroon, Robert I. Davis (Auditory Research Laboratory, SUNY at Plattsburgh, Plattsburgh, NY 12901), and Donald Henderson (Communicative Disorders and Sciences, SUNY at Buffalo, Buffalo, NY 14260)

There is the suggestion in the literature that vibration may potentiate the effects of noise and may pose an increased risk of hearing loss. However, in human experimental studies, which, by necessity, are limited to low levels of TTS, the effects measured are consistent but relatively small. A very limited number of animal studies have also shown an enhanced hearing loss, but the scope of these studies is limited by a large intersubject variability and a small number of subjects. Also, the high levels of stimulation that were used in some of these animal experiments were not realistic. Our recent animal studies (chinchilla) have used a 30 Hz, 3-g-rms cage

vibration in combination with continuous noise (95-dB, 0.5-kHz octave band) and impact noise (113-, 119-, or 125-dB peak SPL) exposure paradigms. All exposures lasted for 5 days. The impact noise exposures were designed to have an equal total energy. Temporary (compound) and permanent threshold shifts were measured using evoked potentials. Sensory cell populations were evaluated with the surface preparation technique. The results obtained from each of the above paradigms were consistent in showing that the presence of vibration did not have a statistically significant effect on hearing thresholds. A parallel set of experiments using a 20 Hz; 2-g-rms vibration is in progress. Preliminary conclusions are essentially the same as those of the 30-Hz experiments. The suitability of the chinchilla as an animal model for use in vibration experiments will also be discussed. [Work supported by NIOSH.]

2:35

**14. Combined effect of whole-body vibration and noise on the dopamine turnover in the rat brain.** Akira Okada, Hiroyuki Nakamura, Hideki Nakamura, and Seiichi Nohara (Department of Public Health, School of Medicine, Kanazawa University, Takaramachi 13-1, Kanazawa 920, Japan)

In order to clarify the combined effects of whole-body vibration and noise on dopamine (DA) metabolism within the brain, which is known to regulate the response of the organism to various stimuli, the DA turnover rate in regions of the rat brain was determined. The rats were divided into five groups: (1) control; (2) whole-body vibration (4 G, 20 Hz, 90 min) exposure alone; (3) noise [70 dB(A), 90 min] exposure alone; (4) noise [100 dB(A), 90 min] exposure alone; and (5) combined exposure for 90 min to whole-body vibration (4 G, 20 Hz) and noise [100 dB(A)]. Changes of plasma corticosterone levels were examined as indices of the pituitary-adrenal function (PAF). The whole-body vibration exposure alone caused increases in the DA turnover rate [an increase of homovanillic acid (HVA) or HVA/DA] in the frontal cortex and nucleus accumbens. Noise exposure alone caused metabolic increases in the amygdala. The combined effect of whole-body vibration and noise on the DA neuron systems suggested that the response of the PAF to the combined stimulus was greater than that to each stimulus alone.

### Contributed Papers

3:05

**15. Results of periodic medical examinations of workers who are exposed to combined noise and vibration.** Klaus Ruppe (Institute of Occupational Medicine, Humboldt-University Berlin, Postbox 140, Berlin 1040, GDR) and Gottfried Enderlein (Central Institute of Occupational Medicine of the GDR, Berlin, GDR)

Synergistic effects of combined exposures to noise and vibration on hearing loss have been recognized under laboratory conditions by some authors. However, there are few studies of the chronic effects of combined workplace factors upon workers. An evaluation of the results of periodic medical examinations of workers who have been exposed to combined exposures during a great part of their working life showed a significant influence of the combination of factors upon hearing loss. In a population of 270 000 male workers, 52.6% were exposed to noise with levels more than 85 dB(A), 15.0% to whole-body vibration (WBV), and 4.1% to hand-arm vibration (HAV). Exposure to noise occurred in 70% of workers also exposed to WBV, and in 80% exposed to HAV. Hearing loss (more than 30 dB in 4 kHz) occurred in 16.7% of the group of workers with exposure to noise alone, but in 18.5% of the group of workers with combined exposure to noise and HAV. The prevalence of medical findings and a restricted capacity to work were significantly higher in the groups with combined exposures.

3:20

**16. Building noise criterion curves, BNC, for interior spaces.** Leo L. Beranek (BBN Laboratories, 10 Moulton Street, Cambridge, MA 02238)

This paper presents an updated set of noise criterion (NC) curves that are renamed building noise criterion (BNC) curves. They are based, in part, on the latest definition of four-band speech interference level and on "spectrum balance," that is to say, the premise that the loudnesses of all bands containing the same number of critical bands should be equal, as

calculated by Stevens Mk. VII perceived loudness method. The curves are extended downward in frequency to include the two octave bands with mean frequencies at 16 and 31.5 Hz. Finally, the high-sound level, low-frequency region of the curves [Blazier, *Noise Control Eng.* 16, 64-73 (1981)] are demarked to show where human annoyance will probably result from vibrations caused by such noise levels in contemporary building construction. The paper details the use of the BNC curves in writing a specification for building construction and in determination of the conformance of the measured result to the specification, or in rating an existing noise. Particularly important are the procedures given for determining "spectrum imbalance." The handling of two typical types of imbalance are discussed: (1) an acceptable speech interference level accompanied by high low-frequency band levels; (2) acceptable low-frequency band levels accompanied by a very low speech interference level.

3:35

**17. Vibrotactile intensity difference thresholds measured by two methods.** George A. Gescheider (Department of Psychology, Hamilton College, Clinton, NY 13323), Stanley J. Bolanowski, Jr. (Department of Physiology, University of Rochester Medical School, Rochester, NY 14642), and Ronald T. Verrillo (Institute for Sensory Research, Syracuse University, Syracuse, NY 13233)

The difference threshold for the detection of changes in vibration amplitude was measured as a function of the intensity and frequency of stimuli delivered through a 2.9-cm<sup>2</sup> contactor to the thenar eminence. Stimuli were either 25- or 250-Hz sinusoids or narrow-band noise centered at 250 Hz or wideband noise. Thresholds were measured by two-interval forced-choice tracking under two methods of stimulus presentation. In the two-burst method, subjects had to judge which of two 700-ms bursts of vibration separated by 1000 ms was more intense. In the increment-detection method, subjects had to detect an increment in the amplitude of vibration. Thresholds were consistently lower for detecting increments in the amplitude of continuous vibration than in detecting amplitude differences be-

tween successive bursts. Amplitude increment detection, however, was relatively poor when the standard stimulus was brief rather than continuous. The near miss to Weber's law was found for both sinusoidal and noise stimuli under both methods of stimulus presentation. The difference threshold was not affected by stimulus frequency. [Work supported by NIH.]

3:50

**18. Determination of natural frequency of bone: Study of bone abnormalities.** Sanjay Yadav and V. R. Singh (Department of Instrumentation and Biomedical Ultrasonics, National Physical Laboratory, New Delhi-110012, India)

Direct and indirect methods have been used in the past [L. Nokes, W. J. Czyz, Mintow, I. Mackie, J. A. Fairclough, and J. Williams, *J. Bio. Med. Eng.* 6, 45-48 (1984)] for the determination of natural frequency of bone. Most of them are cumbersome and costly. A simple and quick experimental technique based on the stress wave propagation through bone is developed. An acoustic vibrator as a source of stress wave is used to send the stress wave through bone specimen under test, in this case, an *in vitro* sample, and a microphone pickup is used to receive the acoustic signal at the other end of the bone, which, in turn, gives an electrical output to be measured on a calibrated electrical detecting system. The natural frequency of bone is determined for an optimum output with respect to different stress wave frequencies. A comparative study of normal and fractured bone is made to diagnose the size of the fracture and its rate of healing.

4:05

**19. Sounds of swallowing.** Sandra Hamlet, Richard Nelson, and Robin Patterson (Department of Otolaryngology, Wayne State University, Detroit, MI 48201)

Sounds of swallowing as detected by a throat microphone have been used in the past primarily to mark the occurrence of swallowing. The source of these sounds and what information the signals might contain about function is relatively unknown. For this investigation, signals from a miniature accelerometer taped to the throat were recorded simulta-

neously with videofluoroscopic data taken while normal subjects swallowed small amounts of liquid barium suspension and barium paste. The progress of the barium "bolus" could thus be followed radiographically, and physical events in swallowing related in time to accelerometer signal characteristics. The most prominent signal feature is a relatively brief (200-ms) broadband noise that corresponds to the rapid passage of the bolus through the lower pharynx and cricopharyngeal sphincter into the esophagus. The spectrum of the noise contains stronger high-frequency components for a liquid than for a paste swallow. In close temporal proximity to this noise component, or even mixed with it, is often a periodic signal that is in the frequency range of high-pitched phonation (approximately 500 Hz) and which may be of laryngeal origin. Other low-amplitude signal features corresponded to structural movement of the hyoid/larynx or epiglottis. [Work supported by NIH.]

4:20

**110. Changes in breath sound spectra with early development of the human respiratory tract.** Elzbieta B. Slawinski (Department of Psychology, The University of Calgary, Calgary, Alberta T2N 1N4, Canada), D. D. McMillan (Department of Pediatrics, The University of Calgary, Calgary, Alberta T2N 1N4, Canada), and D. G. Jamieson (Department of Communicative Disorders, University of Western Ontario, London, Ontario, Canada)

Breath sounds in prematurely born infants are thought to be less "vesicular" than in term infants. This difference in sound quality in newborns of various gestational ages can be explained by different transmission characteristics of lung parenchyma. Chest sounds of 27 infants, of gestational ages between 30 and 40 weeks, were recorded at different chest locations. The results show that there is a strong correlation between chest sounds of normal babies and the location of the recording microphone on the infant's chest. The spectra of inspiration of normal term babies recorded at the level of the lower posterior chest, show dominating amplitudes in the lower frequencies (350 Hz). Sounds recorded at the upper posterior chest, show spectra with enhanced amplitudes in a broader range of higher frequencies (300-900 Hz). The studies also suggest the existence of a marked difference between breathing patterns of prematurely and term born infants. Sounds of premature babies tend to have a lower overall intensity, and spectra with dominant amplitudes in a narrower bandwidth. Moreover, dominant amplitudes are shifted toward higher frequencies as compared to sounds recorded in term infants.

TUESDAY AFTERNOON, 17 MAY 1988

EAST BALLROOM B, 1:00 TO 5:20 P.M.

## Session J. Engineering Acoustics II and Structural Acoustics and Vibration II: Structure-Fluid Interaction Problems

Sung-Hwan Ko, Chairman

Code 2133, Naval Underwater Systems Center, New London, Connecticut 06320

Chairman's Introduction—1:00

### Invited Papers

1:05

**J1. Wave motions in some composite and layered media.** Miguel C. Junger (Cambridge Acoustical Associates, Inc., 54 CambridgePark Drive, Cambridge, MA 02140)

Fluid- and structure-borne waves are governed by the balance between elastic and inertia forces embodied, respectively, in the compliance and density of the wave-bearing medium. This paper is concerned with a class of

composite and layered media where compliance is dominated by one medium and density by another. Examples of this situation are: flexible conduits containing a liquid, e.g., blood vessels; bubble swarms; porous elastic solids with Poisson's ratios in excess of one-third; compliant layers in contact with a liquid, e.g., sedimentary ocean bottom containing gas bubbles or a compliantly coated structure immersed in a liquid.

1:35

**J2. The transient response of fluid-loaded structures using time-domain methods.** Peter R. Stepanishen (Department of Ocean Engineering, University of Rhode Island, Kingston, RI 02881-0814)

Over the past several years, time-domain methods have been developed to address the transient radiation and scattering from structures in fluids with and without a mean flow. These methods are based on using *in vacuo* modal expansions with time-dependent coefficients that are coupled due to the fluid. To date these methods have been used to investigate the transient response of baffled membranes, plates, and cylindrical shells. The development of the basic computational method will be reviewed and compared to alternative methods, e.g., the doubly asymptotic approximation. Numerical results will then be presented to illustrate typical transient phenomena of interest. A particular example of interest is the destabilizing effect of a mean flow over a vibrating surface. [Work supported by the Office of Naval Research.]

2:05

**J3. Measurement of flow excitation function.** Charles H. Sherman, Sung H. Ko, and Barry G. Buehler (Naval Underwater Systems Center, New London, CT 06320)

Wave vector filter measurements have been made with the objective of improving quantitative knowledge of the flow excitation function. Two hydrophone arrays with the same number of elements but different spacings were used. The more closely spaced array covers high wavenumbers characteristic of the convective region of the turbulent boundary layer pressure fluctuations, while the other array covers the lower wavenumber region where contaminants associated with the experimental arrangement also usually exist. The arrays were mounted on a damped composite plate and tested at various flow speeds. Individual hydrophone outputs from both arrays were sampled simultaneously, digitized, recorded, and spectral analyzed in frequency and two wave vector components. Results will be presented as amplitude contour plots as a function of the two wave vector components on the plane of the plate at various frequencies. Results from the closely spaced array show the shape and level of the convective ridge and give an estimate of the difference between the convective peak level and the low wavenumber domain. Results from the other array show that the low wavenumber domain also contains wave vector components associated with free plate vibrations {confirmed by calculations based on work by D. J. Mead and S. Markus [J. Sound Vib. 10(2) (1969)]} as well as acoustic components. [Work supported by ONR.]

2:35-2:50

Break

2:50

**J4. Plates with regular stiffening in acoustic media: Vibration and radiation.** Denys J. Mead (Department of Aeronautics & Astronautics, University of Southampton, Southampton, England)

Flat plates and cylindrical shells with identical stiffeners at regular intervals constitute spatially periodic structures, and specially convenient methods of analysis are available for the study of their vibrations. Some of the methods are suitable for the inclusion of the effects of fluid loading from adjacent acoustic media. This paper outlines the nature of the free wave motion that can occur in periodic structures that are stiffened either in one direction or in two orthogonal directions. It is shown how their responses to distributed sound fields can be determined by using displacement functions consisting of a series of space harmonics or of simple assumed polynomial modes. The sound that is reradiated or transmitted by the structure is also found. Methods that have been developed for analyzing the response and radiation due to line or point forces are reviewed. Recent developments in the analysis of periodically stiffened cylindrical shells are described. The hierarchical finite element method has been applied to determine flexural wave speeds in both flat reinforced plates and in reinforced cylinders. Symbolic computing has been used to set up the relevant stiffness and mass matrices. Some computed results are presented.

3:20

**J5. Control of sound radiation from submerged plates.** Leonard Meirovitch and Surot Thangjitham (Department of Engineering Science and Mechanics, Virginia Polytechnic Institute & State University, Blacksburg, VA 24061)

The subject of sound radiation from submerged elastic bodies, and in particular flat plates, is our current research interest in the field of underwater acoustics. Because the plate is in contact with the surrounding fluid,

it gives rise to a general class of coupled problems involving "radiation loading." This radiation loading modifies the motion of the submerged plate via the excess acoustic pressure, which, in turn, depends on the velocity of the plate. As a result, an interaction between the fluid and the plate exists in the form of "feedback coupling" [M. C. Junger and D. Feit, *Sound, Structures and Their Interaction* (MIT, Cambridge, MA, 1986), 2nd ed.]. In many applications, the radiation pressure is an undesirable effect, so that the object is to suppress it. This paper is concerned with radiation pressure suppression through active control. The general idea of active control of a structure is to suppress the vibration by applying suitable feedback control forces [L. Meirovitch, *Dynamics and Control of Structures* (Wiley, New York, 1989)], which requires sensors capable of measuring the state of the system and actuators capable of applying control forces. A control technique capable of suppressing the farfield radiation pressure is developed.

### Contributed Papers

3:50

**J6. Vibrational response of water-loaded flat plate to a turbulent boundary layer pressure field.** Nilabh Narayan and Robert Plunkett (Department of Aerospace Engineering and Mechanics, University of Minnesota, 110 Union Street SE, Minneapolis, MN 55455)

The vibrational response of a 1.2- × 40- × 200-mm flush-mounted flat plate, with heavy accelerometers attached to the backside, to a turbulent boundary layer (TBL) pressure field was measured at several flow velocities in a 190- × 190-mm low noise water tunnel. The measured response spectra of the water-loaded stainless steel flat plate was compared with the calculated response in the 400- to 1500-Hz frequency range using the plate modal properties and the pressure spectra. The phase indicated convection velocity varied with separation and frequency and was accounted for in the plate response calculations. The plate modal properties were obtained experimentally by *in situ* modal analysis and the plate was found to have a large number of heavily damped modes in the same frequency range. The turbulent boundary layer pressure field driving the plate was measured upstream and downstream from the plate and was found to be unaffected by the vibrating plate. The two-point pressure cross spectra at the wall for the TBL were found to vary with the axial and transverse separations in a way similar to that of the Corcos similarity model and followed the multiplication hypothesis but had a different form of explicit dependence upon the separation.

4:05

**J7. Resonances in the sonar cross sections of shells near the sea surface.** G. C. Gaunaurd (Naval Surface Warfare Center, White Oak Lab (R43), Silver Spring, MD 20903-5000) and M. F. Werby (Naval Ocean Research and Development Activity, Code 221, NSTL Station, MS 39529-5004)

A model for the study of acoustic scattering from a spheroidal elastic shell located near a pressure-release interface has been developed. The model rests on the method of images and on the extended boundary condition (EBC) approach. The method yields a  $T$  matrix that maps the incident field onto the resulting scattered field. The approach is used to describe, compute, and display the echoes backscattered from the shell, as functions of frequency [i.e., the sonar cross section (SCS) of the target] and of angle [i.e., the differential scattering cross section (DSCS)], and it permits the determination of scattering patterns in the appropriate resonance frequency regime. A goal of the work is to assess the influence of a nearby pressure-release boundary on the location of the resonance features in the SCS of submerged *elastic shells*. The model predictions are illustrated with a variety of pertinent numerical displays exhibiting the resonance changes with increasing shell depths. The results are compared

to simpler cases available elsewhere [i.e., IEEE J. Ocean. Eng. OE-12 (special issue on *Scattering*), 380-394, 395-403, and 419-422 (1987)].

4:20

**J8. Noise radiation from transverse air flow over cylinders of constant and varying diameter: Experimental investigation of parameters.** Karen C. Herdman and Paul R. Donovan (Noise and Vibration Laboratory, General Motors Proving Ground, Milford, MI 48042)

Aeolian tones from nonrigid, circular cylinders fixed on one end were measured using sound intensity techniques. For cylinder diameters of the order of 3 to 4 mm and lengths of 750 mm, sound intensity was measured 18 in. from the cylinder axis at a direction perpendicular to the air flow. The radiated sound intensity and Strouhal numbers were determined for several cylinder diameters and flow velocities for the Reynolds number range of 4000 to 11 000. The sound intensity increased at a rate greater than theoretical velocity to the sixth power relationship [N. Curle, Proc. R. Soc. London Ser. A 231, 505-514 (1955)]. The influence of resonant vibration on the radiated sound is examined and compared to previous results [Leehey and Hanson, J. Sound Vib. 13, 465-483 (1971)]. Significant reduction in Aeolian tones with minimal axial taper was demonstrated.

4:35

**J9. Noise radiation from transverse air flow over cylinders of constant and varying diameter: An empirical predictive model.** Paul R. Donovan and Karen C. Herdman (Noise and Vibration Laboratory, General Motors Proving Ground, Milford, MI 48042)

A simplified model to estimate the amplitude and frequency content of the noise radiated by constant and varying (tapered) diameter cylinders has been developed. Using the measured intensity level of Aeolian tones for several constant diameter cylinders, the incremental source strength along the cylinder has been determined as a function of Reynolds numbers. After choosing an element size consistent with the correlation length of the vortex shedding, tapered cylinders are modeled as a series of small, discrete cylinders each of constant diameter. The total sound produced by the tapered cylinder is then determined by summing the contribution of each segment over the cylinder length, properly accounting for geometry using the assumption that the sources add incoherently. The results predicted by this approach have been found to be quantitatively consistent with the data measured for tapered cylinders at several orientations. The model can also be readily extended to examine the effect of nonuniform flow on radiated noise.

**J10. Applications of reactive intensity.** J. Adin Mann, III and Jiri Tichy (Graduate Program in Acoustics, The Pennsylvania State University, P.O. Box 30, State College, PA 16804)

The reactive intensity vector is a very useful analysis tool for the acoustic nearfield. Vibration sources, active and passive absorbers, and diffraction can be differentiated by measuring the reactive intensity. The real and imaginary parts of the specific acoustic impedance should also be considered. Nearfield measurements will be discussed where the reactive intensity vector is used to determine the location and distinguish between piston sources and a resonator that are acoustically coupled. The reactive intensity must be used because the time-averaged intensity vector does not distinguish between a resonator absorbing power and an active source absorbing power. Diffraction will also be identified. These results will further show that all quantities measured with the acoustic intensity technique should be analyzed.

**J11. Water-structure interaction effects in stick-slip bearing vibrations of a submerged frame structure.** Hal Amick (BBN Laboratories, Inc., 21120 Vanowen Street, Canoga Park, CA 91303)

Low-frequency vibrations of a hinged, submerged steel-frame structure were observed during rotation at several speeds. The vibrations had not been observed during operation in air prior to submersion. The vibrations were attributed to stick slip of the submerged hinge bearing caused by high friction. The vibrations were measured and the characteristics of the vibrations appeared to change with rotational speed. A finite element analysis of the gate, which included the "added mass" effects of the water, disclosed that the slip frequencies corresponded to several natural modes of the gate framework. A finite element analysis of the dry structure (without added mass) indicated much higher frequencies for these modes, suggesting that displacements associated with stick slip in the dry structure would not have been large enough to have produced observable vibrations.

TUESDAY AFTERNOON, 17 MAY 1988

GRAND BALLROOM C, 1:00 TO 5:12 P.M.

### Session K. Speech Communication II: Cross-Language Studies: Acoustics, Phonetics, and Perception

James E. Flege, Chairman

*Biocommunications, University of Alabama, University Station, Birmingham, Alabama 35205*

Chairman's Introduction—1:00

#### *Contributed Papers*

1:02

**K1. Voice onset time and  $F_0$  perturbation under conditions of mixed airstream mechanism.** Ian R. A. MacKay and Dariusz A. Zwierzynski (Phonetics Laboratory, Linguistics Department, University of Ottawa, Ottawa, Ontario K1N 6N5, Canada)

Pulmonic and velaric airstream mechanisms are employed together in the production of clicks and accompanying vowels. Questions thus emerge as to the temporal and microprosodic coordination of these two mechanisms, particularly with respect to the phonatory characteristics of clicks. This report is on the acoustic measurement of the post-alveolar click in Sesotho, with particular reference to VOT and  $F_0$  perturbations. There are two phonemically distinct varieties of click in Sesotho, which might be characterized as aspirated and unaspirated in accordance with orthographic conventions. The unaspirated click has a mean VOT of approximately 40 ms on the order of voiceless aspirated plosives in English. The aspirated click has a mean VOT of approximately 120 ms, greater than any plosive type reported by Lisker and Abramson [Language 47, 767-785 (1971)]. The  $F_0$  values were measured pre- and post-click in both word-initial and word-medial clicks. Our data indicate a consistent fall in  $F_0$  immediately prior to a click, regardless of lexical tone, presence of word boundary, and the distinctive prosody of interrogatives. This perturbation of fundamental frequency may be indicative of an intrinsic property of click production. By contrast, measures of post-click  $F_0$  reveal a less regular pattern of microchange both across the silent interval and during the initial glottal cycles at post-click vowel onset; these phenomena are apparently influenced by lexical tone.

1:17

**K2. Interaction effects of nasality and speaking fundamental frequency.** Caroline G. Henton (Linguistics Program, University of California, Davis, Davis, CA 95616)

Previous studies of nasality have concentrated on productional, acoustic, and perceptual features distinguishing nasal segments. Research reported here examines a wider aspect of nasality, namely its interaction with speaking fundamental frequency. To investigate whether nasality lowers or raises SFF, bilingual speakers recorded four passages that differed (a) as to language—French versus English—and (b) as to proportions of nasal segments. Speakers' SFF was measured in the four conditions. Statistical analysis of the results indicates that there is significant interaction of nasality with SFF, with sex of speaker and native language also being important variables. Implications of these results include: first, segmental interaction with prosodic elements; second, in speech synthesis, parameters for nasal segments and  $F_0$  may need to be interdependent, or variable, according to speaker sex or language spoken.

1:32

**K3. Temporal location of syllabic tonal target in standard Chinese.** Zongji Wu and Hongmo Ren (Phonetics Laboratory, Linguistics Institute, Chinese Academy of Social Sciences, Beijing, People's Republic of China)

The understanding of the dynamic nature of the syllable has been deepened by investigating the P center/stress beat and segmental target concentration (Ren, 1986). This study is concerned with another feature, the temporal location of syllabic tonal target in a tone language, Chinese. Chinese has four syllabic tones with the phonological forms H, LH, L, HL, respectively, realized by changes in  $F_0$ . Our acoustic data from three speakers shows that (1) in contrast to the anticipatory cross-syllable coarticulation at the segmental level, the cross-syllable tonal transition is a carryover, (2) with a conflicting preceding tonal context, the temporal location of the only tonal target in H and L and the first target in LH and HL depends on the rate of  $F_0$  transition starting from the syllable onset, (3) with a harmonic preceding tonal context, the first target in LH and

HL is located very close to the consonant offset or the vowel onset. The relations between P center/stress beat and tonal target in case (3) call for further investigation.

1:47

**K4. An acoustic study of tone sandhi in Taiwanese.** Hwei-Bing Lin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695, and Department of Linguistics, University of Connecticut, Storrs, CT 06268)

In Taiwanese, there are five long tones: high level, high falling, mid falling, low rising, and mid level. These tones undergo sandhi changes when they do not precede a major syntactic boundary: mid level → mid falling → high falling → high level → mid level ← low rising. The aim of this study was to validate these phonological observations by investigating the acoustic properties of tones in sequence. Another issue of interest was whether tonal coarticulation occurs in Taiwanese. Six native speakers produced several repetitions of a sequence of two syllables /si/ and /do/ in sentence-final position with all combinations of tones. These utterances were analyzed to determine the average fundamental frequency ( $F_0$ ) contours of the five tones on each syllable. The tonal contours on /do/ syllables were highly similar to those observed previously in isolated syllables. Some slight perseveratory effects of the preceding /si/ tones on the  $F_0$  onset of /do/ tones were found. The tones on the penultimate syllable /si/ underwent sandhi changes, as expected. No coarticulatory effects were found on /si/ tones. However, the sandhi tones differed in their  $F_0$  contours from those on the final /do/ syllable. [Work supported by NICHHD.]

2:02

**K5. Prosodic characteristics and their control in Japanese speech under varying speaking styles.** Masanori Miyatake and Yoshinori Sagisaka (ATR Interpreting Telephony Research Laboratories, Twin 21 Building, MID Tower, 2-1-61 Shiromi Higashi-ku, Osaka 540, Japan)

Two sets of Japanese speech samples were analyzed to clarify the effects of speaking styles on prosodic parameters. The first set of 4068 isolated word utterances consists of 308 different Japanese words uttered in seven different ways, i.e., normal, slow, fast, strong, weak, high, and low. The second set of 110 conversational utterances consists of 11 different Japanese sentences uttered in four different conversational styles, i.e., normal, tender, restless, and irritated, and in one normal reading style. These samples were uttered by two professional narrators and their prosodic parameters [fundamental frequency ( $F_0$ ), power and segmental duration] were compared. The following tendencies were found from the analysis: (1) For most ordinary speaking styles (all but "high")  $F_0$  and power are varied together. Correlations of 0.81–0.99 were measured in these cases. (2)  $F_0$  can, however, be controlled independent of power, when subjects were instructed to do so (e.g., speaking style: high). (3) The average values and the dynamic ranges of  $F_0$  patterns differ between speaking styles. To realize these tendencies for speech synthesis by rule, a  $F_0$  control model is proposed. Applying analysis by synthesis to all samples in the first set at a time, the system parameters of this model could be derived. It is confirmed that the  $F_0$  pattern can be systematically generated by the proposed model for various speaking styles.

2:17

**K6. Cross-language perception of a place-of-articulation contrast: The influence of nonspecific linguistic experience.** Linda Polka (Department of Psychology, University of South Florida, Tampa, FL 33620)

There is some debate whether the effects of experience on perception of nonnative speech contrasts reflect psychoacoustic, phonetic, or phonemic factors. The present study addresses this issue by comparing the performance of native English speakers and native speakers of Farsi (Per-

sian) on perceptual categorization of ejective uvular and velar stops in Thompson (a North American Indian language). While the ejective manner class is not represented in the phonetic inventory of either English or Farsi, the uvular-velar distinction has a different phonological status in the two languages. In English, the uvular-velar place contrast is not a phonemic distinction. In Farsi, this place distinction is phonemic for voiced (nonejective) stops. To evaluate the effect of this phonological status difference, perceptual categorization of multiple natural exemplars of Thompson /k/ and /q/ (produced in the same vowel context) was assessed in both language groups. Subjects completed both an identification test and a name-identity (categorical) same-different discrimination test. Results are discussed as they address the role of phonetic (articulatory feature similarities between native and nonnative phonologies in perception of nonnative speech contrasts. [Research supported by NINCDS and NICHHD.]

2:32

**K7. Perceptual switching in Spanish/English bilinguals.** O. S. Bohn and J. E. Flege (Biocommunication, University of Alabama in Birmingham, Birmingham, AL 35294)

Elman *et al.* [J. Acoust. Soc. Am. 62, 971–974 (1971)] found that, whereas English monolinguals heard short-lag stops as /b/, Spanish monolinguals heard them as /p/. Spanish/English bilinguals who were proficient in English gave more "English" (i.e., /b/) responses in an English than Spanish perceptual "set." This study further tested the hypothesis that bilinguals may use two criteria in identifying short-lag stops. In experiment 1, /to/ tokens spoken by Spanish monolinguals with short-lag VOT were presented along with English long-lag /to/ or English short-lag /do/ tokens. English monolinguals and Spanish subjects who learned English as adults labeled the Spanish /to/ stimuli as /to/ (rather than /do/) more often in the context of English /do/ (73%) than /to/ (50%). Since both groups showed the same pattern, this likely reflected a context effect rather than a criterion shift. In experiment 2, the English monolinguals did not label the Spanish /to/ stimuli as /to/ more often in a Spanish than English "set." Whereas the bilinguals who participated in the Spanish set first showed a 12% shift, those who participated in the English set first showed no evidence of a switch in criteria. It has been hypothesized [Flege and Eefting, J. Acoust. Soc. Am. 83, 729–740 (1988)] that Spanish speakers who learn English in early childhood establish separate categories for Spanish and English /t/. Work is underway to determine if such subjects will show a larger language set effect than the adult L2 learners already examined. [Work supported by NIH.]

2:47

**K8. Native and nonnative perception of voicing in English.** Joann Fokes (School of Hearing and Speech Sciences Ohio University, Athens, OH 45701) and Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701)

Naturally produced stop-vowel syllables (pea/bee, pie/bye, tea/dee, tie/dye) were selected to represent various voice timing values from the prevoiced, unaspirated, and aspirated ranges. The set of tokens included some intended voiced and intended voiceless stops produced with the same voice onset timing (VOT) values within the range  $+/- 15$  ms. Native and nonnative English speakers identified three repetitions of each syllable in random order for a total of 96 test items. As expected, American English listeners judged all values in excess of 40 ms. VOT as voiceless, and all other values as voiced. In almost all cases, nonnative listener responses were predictable on the basis of their native language. Since Chinese and Japanese contrast stops on the basis of aspiration, listener judgments were almost identical to the American. On the other hand, Malaysian listeners identified all voiceless and voiced stops as intended by the speakers; that is, listeners based voicing judgments on other properties of the syllables than voice timing values. The Arab judgments were almost identical to the American for the alveolar stops /t,d/ but showed considerable uncertainty for the labial stops /p,b/. Bengali speakers also followed the American voicing pattern but showed some confusion for words containing the diphthong /ai/.



3:12

**K9. Voicing in French and English labial stops: A cross-language perceptual study.** Murray J. Munro and Terrance M. Nearey (Department of Linguistics, University of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Measurements of European French and Canadian English labial stops indicate that the French /p/ and English /b/ categories are very similar, often exhibiting only small differences in voice onset time. Two groups of English-speaking listeners (one with some knowledge of French, the other without) were asked to categorize a set of modified natural tokens from these categories as either /p/ or /b/. The tokens were taken from word-initial stops produced in a sentence context. Components preceding the release bursts were removed, and the signals were truncated at 68 ms. Despite the small difference in VOT between the categories, most listeners were able to reliably separate them at levels above chance. Analysis revealed that the listeners may have relied primarily on VOT in making their judgments and that overall amplitude [C. J. Darwin and M. Pearson, *Speech Commun.* 1, 29-44 (1982)] may have played a secondary role. These findings indicate that listeners may be sensitive to small differences between categories in their native language and analogous categories in a foreign language.

3:27

**K10. Acoustic characteristics of tongue root position and vowel assimilation in Akan.** Susan Hess (Department of Linguistics, UCLA, 405 Hilgard Avenue, Los Angeles, CA 90024)

The purpose of this research was twofold: (1) To find an acoustic measure (or diagnosis) of tongue root position in a language where a difference of tongue root position is systematic and significant, and (2) apply this measure to cases of vowel assimilation between vowels of conflicting tongue root positions in order to determine whether tongue root positions were affected. Akan, a Kwa language spoken in Ghana, exhibits a form of vowel harmony controlled by the tongue root in which the vowels in a particular word must be spoken with either an advanced or a retracted tongue root. The position of the tongue root [i.e., either advanced or retracted; hereafter referred to as either + or - ATR (advanced tongue root)] is determined by the word stem, and all affixes must agree with that tongue root position. In this paper, we will look at data from Kwawu, an Akan dialect, which has the following vowels: [i, e, u, o] are [+ATR] vowels, while [ɛ, ɔ, ɔ, a] are [-ATR] vowels. All [-ATR] vowels have variants when they occur in a syllable that precedes another word containing the high [+ATR] vowels [i, u]. Two claims have been made about these variants. One asserts that the variants assimilate in tongue root position to the following [+ATR] vowel and become [+ATR] vowels themselves, while the other argues that the assimilation does not involve the tongue root but merely raising of the tongue blade. Resolving this dispute required that we first define a measure that would provide a reliable diagnosis of tongue root position. Only acoustic measures were considered as other means were not practical. Five kinds of measures were examined: formant frequency, formant bandwidth, duration, pitch, and amplitude of the first three harmonics. It was found that frequency bandwidth (more precisely the bandwidth of the first formant) provided the only reliable diagnosis of vowel type. The [+ATR] vowels were found to have a greatly narrower first formant when compared with [-ATR] vowels. This measure was then used to examine the behavior of one of the [-ATR] vowels, the mid vowel [ɛ], in two contexts: before the high [+ATR] vowels [i, u], and before the low [-ATR] vowel [a]. Bandwidth measurements of ɛ before the [+ATR] vowels [i, u] do not show the narrower bandwidth found in [+ATR] vowels, allowing us to preliminarily conclude that tongue root position is not assimilated.

3:42

**K11. Formant onset duration: A secondary cue for identifying the place of articulation in stop consonants.** H. Garudadri, M. P. Beddoes (Electrical Engineering, University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada), J. H. V. Gilbert, and A. P. Benguerel (Audiology and Speech Sciences, University of British Columbia, Vancouver, British Columbia V6T 1W5, Canada)

Formant onset duration (FOD) is suggested as a possible alternative to voice onset time (VOT), to characterize the place of articulation in stop consonants. The FOD is defined as the duration from the start of the burst (denoting consonantal release) to the start of the formant structure (denoting vowel onset). In many languages, VOT is the most salient cue distinguishing between voiced and voiceless stop phonemes. Although VOT has been shown to increase in duration from bilabials to alveolars to velars [Lisker and Abramson, *Word* 20; Zue, *Tech. Rep.* 523, MIT], attempts to use VOT as a cue to place of articulation [Kewley-Port, *J. Acoust. Soc. Am.* 73, 322-335 (1983)] were not successful. In a recent study of stop consonants using the partially smoothed Wigner distribution [Garudadri *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 82, S55 (1987)], FOD and  $F_2$  of the following vowel were used along with the spectral shape during the burst, to resolve context dependencies due to coarticulation effects. The FOD increases from bilabials to alveolars to velars and appears to be related to the inertia of the articulators. Unlike VOT, FOD is always positive and corresponds to the vocal tract rather than the source of excitation. Evidence from English, Telugu, and French stop consonants supporting these ideas is presented. [Work supported in part by NSERC, Canada.]

3:57

**K12. Prenasalized stops and speech timing.** Ian Maddieson (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024)

Prenasalized stops raise a number of interesting questions concerning the relationship between phonological units and timing in speech. Fijian is a language in which prenasalized stops are incontrovertibly single segments from the phonological point of view. It has two stop series, voiced prenasalized and voiceless oral. All syllables are open and no consonant clusters occur. Hence, this is a particularly appropriate language in which to examine these questions. Utterance-initial and word-medial prenasalized stops at bilabial, alveolar, post-alveolar, and velar places of articulation were examined in a set of words spoken by 11 speakers representing three main varieties of Fijian. Acoustic duration was measured from onset of nasal murmur to stop release (the stop component of a prenasalized stop is usually very brief and was not measured separately). Initial prenasalized stops are significantly shorter than medial ones. Medial prenasalized stops are comparable in duration to intervocalic laterals and voiceless stops, but initial prenasalized stops are significantly longer than the initial nasals in unstressed syllables which occur in the data set. Vowel length before prenasalized stops is comparable to that before other consonants. These results indicate that the role of prenasalized stops in the timing pattern of Fijian is generally similar to that of other single segments. In particular, they do not act like a consonant sequence.

4:12

**K13. Differences in relative temporal positioning between initial consonant and vowel in syllables: Evidence from Chinese.** Hongmo Ren (Phonetics Laboratory, Linguistics Institute, Chinese Academy of Social Sciences, Beijing, People's Republic of China)

Junctures and coarticulations between syllable initial consonant and vowel have usually been described in terms of transitions and target varia-

tions. This study proposes that we also need to specify differences in relative temporal positioning between these two major syllabic constituents. The *C* + /iau/ syllables in Chinese were examined. The /iau/ is a triphthong rhyme. Because rhyme internal formant pattern remains constant in different syllables, the turning point in formant trajectories can be used as reference for determining the temporal position of the rhyme as a whole. In fact, we have observed the whole rhyme shifting along the time axis relative to the initial consonant boundary. Data from six speakers shows that syllable initial consonant and rhyme exhibit different degrees of overlapping and "repelling," with a distance ranging from -15 to 83 ms. The relation between these findings and the P center or stress beat measured in the same tokens is also discussed. [Work supported by NSF.]

4:27

**K14. Analysis and prediction of vowel devocalization in isolated Japanese words.** Hisao Kuwabara and Kazuya Takeda (ATR Interpreting Telephony Research Laboratories, Twin 21 Building MID Tower, 2-1-61 Shiromi, Higashi-ku, Osaka 540, Japan)

An analysis of vowel devocalization in Japanese is presented as an example to find general rules for allophonic variations of phonemes in continuous speech, i.e., to find quantitative relations between phoneme environments and allophonic variations. A speech database consisting of about 5000 common Japanese words spoken in isolation by a professional male announcer was used. The database used is part of a large scale speech database that is now being compiled and hand transcribed at several levels of phonetic detail. Based on the acoustic-phonetic transcriptions of the speech data, factor analysis of the phonetic environment was performed for the devocalized vowels. The results reveal that the influence of the immediately following phoneme on devocalization is the greatest among the four factors examined. The second most important factor is the position of the accent nucleus. The automatic prediction of vowel devocalization is discussed using the results of this analysis.

4:42

**K15. Pitch neutralization in Japanese one mora homonyms.** Yuko Mogami (1431-1 Toba Akashi, 673, Japan)

Japanese one mora homonyms in phrases are distinguished by pitch.

The present experiment was designed to discover if this distinction between the homonyms was also retained in an isolated environment; i.e., if speakers distinctively pronounced the homonyms in isolation and if listeners could correctly identify them. Following Fourakis and Iverson (1984), the influence of the speakers' attention to the purpose of the experiment was considered in the analysis of the experiment. Four pairs of homonyms, of presumably equal frequency in use, were recorded in two ways: first, subjects unaware of the homonyms orally responded to questions whose answers contained the homonyms as well as other decoy words; second, subjects orally read a list of the same homonyms. The obtained homonyms were compared with respect to  $F_0$  value, duration, and pitch contour and then used for listeners' identification task. Both the acoustical analysis and the results of the listeners' identification task proved that one mora homonyms were completely neutralized in an isolated environment regardless of the speakers' attention to the homonyms. [Special thanks to John Ohala, Masayoshi Shibatani, Yoko Hasegawa, and Mariscela Amador.]

4:57

**K16. Delayed pitch fall in Japanese.** Yoko Hasegawa (Department of Linguistics, University of California, Berkeley, CA 94720) and Kazue Hata (Speech Technology Laboratory, Santa Barbara, CA 93105)

The Tokyo dialect of Japanese is regarded as a prototypical pitch accent language where the location of  $F_0$  fall is the only acoustic correlate of the accent. Neustupný [Onsei-gakkai Kaihoo 121 (1966)], however, reported that  $F_0$  fall does not always synchronize with an accented mora but may be carried over to the following mora. Sugito [Shooin Joshi Daigaku Ronshuu 10 (1972)] found that native speakers perceived a pitch accent on a mora when it was followed by a falling  $F_0$  contour and also that the  $F_0$  peak value in the accented mora need not be higher than that in the following mora. Not well investigated though is the correlation between the peak location and the steepness of falling contour. The working hypothesis is that the later the  $F_0$  fall occurs, the faster  $F_0$  drops. This delayed pitch fall is characterized in terms of (1) the  $F_0$  peak location and (2) the steepness of  $F_0$  contour computed in Hz/cs. The results from 560 tokens uttered by seven speakers of the Tokyo dialect confirmed our hypothesis. [Work partially supported by Sloan grant to Berkeley Cognitive Science Program.]

TUESDAY AFTERNOON, 17 MAY 1988

ASPEN ROOM, 1:30 TO 4:50 P.M.

### Session L. Musical Acoustics I: Synthesis of the Singing Voice

Ingo R. Titze, Chairman

*Department of Speech Pathology and Audiology, University of Iowa, 127A SHC, Iowa City, Iowa 52242*

Chairman's Introduction—1:30

#### *Invited Papers*

1:35

**L1. Computer generation of singing by manipulation of speech parameters.** Joseph P. Olive (Linguistics Department, AT&T Bell Laboratories, Murray Hill, NJ 07974)

This paper describes a computer system for simulating a singing voice by using parameters derived from natural speech. The system was constructed to manipulate parametrized speech to study prosodic and segmental properties of speech. The system enabled a user to manipulate fundamental frequency, amplitude, timing,

and spectral characteristics of speech sounds. While manipulations of the parameters of natural speech could easily produce "unreal" or "weird" sounds, simulating a singing voice with qualities approaching those of a natural singing voice was more difficult. The latter required careful manipulations of the fundamental frequency and timing to imitate human singers. Examples of both types of sounds will be demonstrated. In an attempt to imitate natural singing, the simulation of solo and choral singing, as well as contrasting singing styles of "classical" and "pop" singers is described.

2:05

**L2. Singing synthesis with models of phonatory and articulatory kinematics.** Ingo R. Titze (Voice Acoustics and Biomechanics Laboratory, Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242 and Recording and Research Center, the Denver Center for the Performing Arts, Denver, CO 80202)

Singing synthesis to date has been attempted primarily in the acoustic domain. In an effort to learn more about the interactions between the source and subglottal and supraglottal resonators, a simulation strategy is employed that computes pressures and flows on the basis of vocal fold and articulatory movement. Some aspects of the perception of vocal registers are shown to be related to subglottal resonances. In particular, the first subglottal formant (ca. 500 Hz) helps to trigger the involuntary chest-falsetto transition in untrained voices. Roundness in the medial surface contour of the vocal folds also helps to distinguish chest-falsetto and male-female differences. Vocal tract constrictions in the vicinity of the glottis are shown to affect vocal quality, in particular, the perception of vocal ring. Frication in the glottis and the vocal tract is computed on the basis of critical Reynolds numbers and is an automatic consequence of changes in subglottal pressure and airway constriction. Some examples of simulated sound will be given. [Work supported by NIH, Grant No. NS 16320-08.]

2:35

**L3. Analysis and synthesis of the singing voice and musical applications.** Xavier Rodet (IRCAM 31, rue Saint-Merri, 75004 Paris, France)

IRCAM (Institut de Recherche et de Coordination Acoustique/Musique) has been involved in sound analysis and synthesis for contemporary music creation for many years. Accurate analysis and high-quality sound synthesis are thus extensively studied. Some of the methods that have been developed are described with particular attention to the singing voice and speech. A power spectral density estimate has been developed of the speech signal, which is simultaneously very precise and computed on very short segments of the signal in order to follow rapid transitions such as those found in plosives. Spectral envelopes are represented in a flexible way: They are coded in terms of frequency, amplitude, and bandwidth of pairs of conjugate poles (*formants*). An algorithm has been developed giving an harmonic/noisy decision in narrow frequency bands covering the entire spectrum. This technique has greatly improved the sound quality. Several synthesis methods have been developed and compared, including formant waveform synthesis (analogous to parallel formant synthesis), additive synthesis, and serial filter synthesis. Two types of synthesis rules have been studied. On the one hand, rules concerning phoneme coarticulation using *formant* coded diphones, sophisticated algorithm smoothing formant trajectories by taking into account three successive phonemes, durations of phonemes, speed of singing, speed of transitions, and relative timing of formant trajectories. On the other hand, a set of rules has been built with musical performance. Those rules calculate such features as vibrato excursion and rate, vocal effort, and formant locations according to fundamental frequency. Examples of synthesis by rule are given showing different types of voices and styles plus excerpts from concert pieces.

3:05

**L4. Singing synthesis and composition.** Gerald Bennett (Swiss Center for Computer Music, Oetwil am See, Switzerland)

Apart from its interest as a confirmation of experimental hypotheses, singing synthesis can be of great importance in contemporary composition. The experience of many composers with the program CHANP, developed at IRCAM, shows that a sound synthesis program based on a specific physical model can serve as a general purpose synthesis tool, while remaining intuitively manageable. The particular structure of the program CHANP offers the possibility of defining both the local structure of the sound itself and the larger scale structure of the composition. Musical examples will illustrate both vocal and nonvocal sounds taken from electroacoustic compositions.

3:35

**L5. Synthesizing singing.** Johan Sundberg (Department of Speech Communication and Music Acoustics, Royal Institute of Technology [KTH], S-100 44, Stockholm, Sweden)

An attempt to synthesize singing is presented in which a synthesizer containing five cascaded formants is complemented by two noise generators and fixed filters. The performance is controlled by computers reading

the input notation and processing it with pronunciation and musical performance rules. Both solo and choral singing are synthesized. Various effects will be demonstrated, such as timing of pitch changes and consonants, coloratura, markato, timbre, legato, and multiple pitch singing. Also, some effects concerning musical expression will be presented.

### Contributed Papers

4:05

**L6. Is there a single vibrato waveform?** Robert Maher and James Beauchamp (Computer Music Project, School of Music, University of Illinois at Urbana-Champaign, 2136 Music Building, 1114 W. Nevada, Urbana, IL 61801)

High-fidelity singing synthesis requires careful consideration of the properties and character of natural vibrato. In many synthesis methods, a single vibrato waveform is derived and applied equally to all of the partials of the synthesized tone. The research reported here tests the validity of this linear treatment of vibrato using an analysis/synthesis procedure. The singing tone /ah/ is analyzed for bass, tenor, alto, and soprano voices, providing time-variant measurements of amplitude, frequency, and phase for each of the partials. The tones are reconstructed using an additive synthesis procedure. The importance of vibrato waveform similarities and differences between partials is evaluated by resynthesis of the singing tones from modified analysis data. The modifications include the use of the fundamental partial's vibrato waveform for all of the partials, the use of sinusoidal vibrato for all the partials, the elimination of amplitude variations for each partial, and various combinations. The analysis data and examples of the resynthesis will be presented. [Work supported, in part, by an NSF Graduate Fellowship.]

4:20

**L7. The jnd's for rate and extent of frequency modulations.** Yoshiyuki Horii and Ron Scherer (Campus Box 409, CDSS, The University of Colorado, Boulder, CO 80309 and The Denver Center for the Performing Arts, Denver, CO 80202)

Vocal vibrato samples were spectrographically analyzed to determine ranges of rate and extent of frequency modulations. Subsequently, sinusoidal and pulse signals of 2-s duration at 261 or 522 Hz, modulated within the measured ranges, were used as standard stimuli to derive rate and extent jnd. For the rate jnd, the standard stimuli were sinusoidally modulated at 4, 5, and 6 Hz. For the frequency extent jnd, the standard

stimuli were sinusoidally modulated at 5 Hz with frequency extent of 0.25, 0.5, and 0.75 semitones. Subjects were instructed to adjust rate/extent of frequency modulations until the difference between the standard and adjustable signals was just noticeable. The rate jnd was less than 5%, while frequency extent jnd was much smaller especially for the pulse stimuli. A perceptual cue for the frequency extent appeared to be rate of frequency changes rather than the extreme frequency values. A follow-up experiment indicated listeners' inability to match accurately nonmodulated signal frequency to the extreme frequency values of the modulated signals. Implications of these findings to vibrato singing will be discussed. [Work supported by NIH.]

4:35

**L8. Analysis, resynthesis, and modification of sound signals with the help of wavelet transforms (i.e., time-and-scale representations).** R. Kronland-Martinet (Laboratoire de Mécanique et d'Acoustique, C.N.R.S., 31, chemin J. Alguier, 13402 Marseille Cedex 9, France) and A. Grossmann (Centre de Physique Théorique, C.N.R.S.-Luminy Case 907, 13288 Marseille Cedex 9, France)

An attempt will be made to present an up-to-date survey of results on wavelet transforms of signals, with particular emphasis on examples involving sound. The main feature of a wavelet transform is that it associates to a signal  $s(t)$  an appropriate function  $S(b,a)$  of two parameters: a time  $b$  and a scale parameter  $a$ . If it is required that the correspondence  $s \rightarrow S$  be linear, the definition of a transform involves an "analyzing wavelet"  $g$ , which has to satisfy some conditions, and, given by  $S(b,a) = Ka^{-1/2} \times \int s(t)g^-[(t-b)/a]dt$ ,  $g^-$  is the complex conjugate of  $g$ . This correspondence has a stable inverse. A transform  $S(b,a)$  is fully determined by its values on a suitable discrete grid of the  $(b,a)$  open half-plane. Wavelet transforms are close relatives of time-frequency representations, the frequency being replaced by a scale parameter. The aim will be to describe some of the mathematical background, to give a description of some existing implementations, and to display an appropriate set of examples, involving acoustical signatures in the field of speech and music. There will be auditive examples showing the possibilities of intimate modifications of sound by these methods.

TUESDAY AFTERNOON, 17 MAY 1988

GRAND BALLROOM A, 1:30 TO 4:20 P.M.

### Session M. Noise III: Airport Noise Compatibility Planning

Dennis G. Ossenkop, Chairman

Federal Aviation Administration, 17900 Pacific Highway South, Seattle, Washington 98056

Chairman's Introduction—1:30

#### Invited Papers

1:35

**M1. Airport noise compatibility planning perspectives.** Steven Starley (Noise Abatement Division, Federal Aviation Administration, 800 Independence Avenue, S. W., Washington, DC 20591)

Air transportation is mass transportation and its continuing growth has spawned a complex set of interrelated noise and capacity problems. Airport proprietors, who are responsible for noise damages, are turning to

the use of restrictions to limit noise in adjacent communities. Use restrictions at one airport affect system capacity and operations at other airports. When capacity limitations affect interstate commerce, airports are caught between their adjacent communities and the Federal government that cannot allow an undue burden on interstate commerce. Airport noise abatement planning, which is effective for airport specific problems, may be a factor in the growth of national problems. Thus a national perspective must be applied to the solution of local problems. This paper traces the development of airport noise abatement planning, discusses some newly emerging areas of interest at both the local and national level, and presents a glimpse of some alternative futures.

2:05

**M2. Attenuating jet aircraft noise in single-family residences near major U.S. airports.** Barney Myer (Facilities and Real Estate Department, Port of Seattle, SeaTac International Airport, P.O. Box 68727, Seattle, WA 98168)

Most noise attenuation of single-family residences near major U.S. airports has been achieved by acquisition and removal of homes. This noise remedy is not only extremely costly but it also disrupts social networks, undermines education and social institutions, lowers tax bases, and can generate subsequent land use problems. Acquisition as a noise remedy is limited to those areas where jet noise is the worst. The thousands of homes that cannot be purchased but endure noise peaks of near 100 dB must be addressed by other remedies. SeaTac International Airport and other airports are initiating acoustical insulation programs to lower interior noise levels to acceptable levels. An issue involved in these developing noise mitigation programs is how to set and achieve noise level goals that realistically make interior noise levels livable *and* are acceptable to the Federal Aviation Administration that funds up to 80% of program costs. The issue involves answering questions such as: Are interior peak noise levels (SEL) to be reduced low enough to allow uninterrupted communication (i.e., TV, telephone, conversation)? Is an average noise criterion ( $L_{dn}$ ) sufficient? Is it possible to standardize interior noise goals when airports have widely divergent levels of activity? Noise remedy programs at SeaTac and other airports must address this basic issue to assure continued funding of their multimillion dollar efforts.

2:35

**M3. The cooperative approach to noise compatibility planning at Portland International Airport.** John P. Newell (Port of Portland, P.O. Box 3529, Portland, OR 97208)

The Port of Portland's cooperative approach to airport noise compatibility planning has produced a plan that seeks to balance the concerns of the community with the needs of the airport and its users. This plan has reduced the number of residents within the  $L_{dn}$  55 contour by 85%, without employing costly operating restrictions. The success achieved is directly attributable to the cooperative process that involves the utilization of an advisory committee to help identify and resolve airport noise issues. This committee, consisting of all parties having an interest in the airport noise issue, has worked together since 1982 in structuring and implementing the noise abatement plan. This plan is made up of three interrelated programs: an operational program, a land use program, and a review and monitoring program. The operational program includes specific flight track assignments, while the land use program requires sound insulation of some residential dwellings, and adoption of city and county noise overlay zones. The review and monitoring program is the part of the plan that insures the ongoing cooperative process, and maintains compatibility with the community.

3:05

**M4. The use of local opportunities in aircraft noise abatement planning.** Jon M. Woodward (Coffman Associates, Inc., 1200 E. 104th Street, Suite 350, Kansas City, MO 64131)

Aircraft noise abatement planning often overlooks low-cost, easy to implement solutions that take advantage of unique local conditions. The focus is too often on generic "cookbook" solutions that are applied to every airport in the same way, e.g., curfews, aircraft type restrictions, or noise budgets. For every airport, careful analysis will reveal unique conditions that provide opportunities to abate noise. These include airport operating schemes such as preferential runway use programs, rerouting of traffic along corridors of compatible land uses, and the use of comparative aircraft versus background noise levels to direct traffic flows. Preferred aircraft operating procedures are selected based upon local underlying surface conditions of land use or terrain. Limitations on aircraft operation through imposition of local legislation can result in noise reduction in specialized cases. Construction of new facilities most often is driven by capacity needs as well as noise abatement. Examples of the effects of using local opportunities for noise abatement are presented for each type of measure from airport noise compatibility planning projects at Portland, Seattle, Phoenix, Anchorage, Indianapolis, Baton Rouge, New Orleans, Chicago, Omaha, Little Rock, and Reno.

**M5. Use of seasonal  $L_{dn}$  for airport noise compatibility planning.** Andrew S. Harris (Harris Miller Miller & Hanson Inc., 429 Marrett Road, Lexington, MA 02173)

The differences between  $L_{dn}$  from yearly average operations and  $L_{dn}$  from busy day or seasonal operations are examined. Civil and military airports are considered. For civil airports, the issue of seasonality is addressed as it relates to the Federal Aviation Administration's requirements under FAR Part 150. For both civil and military airports, current examples are discussed. The environments of civil and military airports are also examined in the context of EPA Task Group 3's recommendations about seasonality ["Impact Characterization of Noise Including Implications of Identifying and Achieving Levels of Cumulative Noise Exposure," for Environmental Protection Agency Aircraft/Airport Noise Report Study, Task Group 3, Henning von Gierke, Chairman, Washington, 1 June 1973.] Methods for developing seasonal noise exposure are examined.

### Contributed Paper

4:05

**M6. Measurement and analysis techniques for determining acoustical impacts from enroute aircraft in very quiet settings.** Robin T. Harrison (1275 North Indian Hill Boulevard, Claremont, CA 91711) and Paul H. Dunholter (Mestre Greve Associates, 280 Newport Center Drive, Suite 230, Newport Beach, CA 92660)

The National Parks of the United States have been set aside for the public enjoyment of pristine settings with minimal impacts from common urban disturbances. Ambient sound levels in these settings are generally very low, and have been measured below 20 dBA. Aircraft noise from enroute high altitude aircraft and sight-seeing aircraft have become a

source of disturbance to park visitors. Under such quiet background conditions, enroute aircraft are often audible for extended durations. In the National Park setting, it has been found that even the detection of intrusive sound can result in annoyance. Numerous and continuing complaints have been generated by overflights that would not be expected to have a significant impact based upon traditional aircraft noise assessment criteria alone. This paper summarizes a National Park Service funded study that addresses various techniques for the measurement of enroute aircraft noise in these quiet settings. These techniques include A-weighted and detectability metrics for both the ambient and aircraft noise. Methods including the effects of the duration of the overflight and the number of events are also reviewed.

TUESDAY AFTERNOON, 17 MAY 1988

GRAND BALLROOM B, 1:30 TO 4:30 P.M.

## Session N. Psychological Acoustics II: Masking, Modulation and Modulation Masking

Christopher W. Turner, Chairman

*Department of Communicative Disorders, Syracuse University, 805 S. Crouse Avenue, Syracuse, New York 13210*

### Contributed Papers

1:30

**N1. Frequency selectivity in focused attention as demonstrated by a multiprobe method.** Huanping Dai and Bertram Scharf (Auditory Perception Laboratory, 413 MU, Northeastern University, Boston, MA 02115)

A multiprobe paradigm, patterned after the probe-signal method of Greenberg and Larkin [J. Acoust. Soc. Am. **44**, 1513-1523 (1968)], led observers to focus attention on a primary frequency in a detection task (2IFC). In each block of 64 trials, the signal was presented 48 times at the primary frequency and twice at each of eight probe frequencies, four higher and four lower than the primary. By presenting many probes in a block instead of only one as in most previous work, across-block variability was avoided. It also made it unlikely that observers focused on frequencies other than the primary. Results from four observers tested in separate sessions on primaries at 250, 500, 1000, and 2000 Hz revealed a consistent drop from 90% detection of the primary to chance detection of probes more than half a critical band away from the primary. At a primary of 4000 Hz, three observers showed the same kind of selectivity, but two others showed less. Generally, informed and naive observers performed much the same way. Since practice also mattered little, a sharp filterlike function could be measured in under 1000 trials. [Work supported by NIH.]

1:45

**N2. Combined masking under conditions of high uncertainty.** Donna L. Neff, Walt Jesteadt, and Brian P. Callaghan (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131)

Simultaneous maskers composed of a few sinusoids scattered over a wide frequency range produce large amounts of masking if the frequency composition of the maskers is changed with each stimulus presentation. This masking appears to be produced primarily by more central processes driven by stimulus uncertainty. In this study, the masking produced by combining these maskers with broadband noise is examined. Growth of masking functions for broadband noise and for multicomponent maskers with 2, 6, 10, 50, and 100 components were used to select levels for individual maskers that produced 10, 20, 30, or 40 dB of masking. Multicomponent and noise maskers were then presented in equated and unequated combinations. The average amount of additional masking beyond a power summation measured for equated conditions decreased from around 12 to 2 dB as the number of components in the multicomponent masker increased from 2 to 100. The data are well fitted by Lutfi's model for combined simultaneous maskers, with exponents that approach 1.0 as the number of components increases to more closely approximate broadband noise. [Work supported by NIH and AFOSR.]

2:00

**N3. Changes in the masked thresholds of brief tones produced by preceding bursts of bandpass and notched noise.** Robert P. Carlyon (MRC Institute of Hearing Research, University Park, Nottingham NG7 2RD, England)

Thresholds were measured for 5-ms 1-kHz tones masked by synchronous bursts of noise containing a spectral notch centered on the signal frequency. These thresholds were reduced by prior exposure to a 200-ms burst of a "priming stimulus," which had the same spectral shape as the masker. This masking release was greatest for notch widths extending between 20%–30% on either side of the signal frequency, and was absent when the masker and primer contained no notch. A smaller masking release could be obtained with primers consisting of only the lower band of a notched noise masker, and, to a lesser extent, of the higher band alone. A primer consisting of a narrow band of noise centered on the signal frequency produced an *increase* in masking, which could not be attributed to forward masking of the tone. The effects of all of these primers were independent of the 30-dB range of primer levels studied, ruling out an explanation in terms of peripheral adaptation. The results are consistent with the presence of an active neural mechanism that enhances the internal representation of newly arriving energy in a previously unstimulated frequency region. Temporal parameters of the masking release were also studied.

2:15

**N4. Comodulation masking release with delayed signal onsets.** Beverly A. Wright and Dennis McFadden (Department of Psychology, University of Texas, Austin, TX 78712)

The detectability of a 1250-Hz, 50-ms signal was assessed in a comodulation masking release (CMR) setting. One 50-Hz noise band (the "masker") was centered at 1250 Hz; other noise bands were centered at 850, 1050, 1450, and 1650 Hz (the "cue" bands). The masker and/or cue bands were gated prior to the onset of the signal by an amount ("fringe") that was varied across blocks of trials. The noise bands and the signal always ended together. The temporal envelopes of the noise bands were all correlated, all correlated except for the masker band, or all uncorrelated. When all of the noise bands were gated synchronously, the CMR grew

3:00

**N6. Signal threshold as a function of the relative modulation depth between on-frequency and flanking masker components.** John H. Grose and Joseph W. Hall, III (Division of Otolaryngology Head and Neck Surgery, School of Medicine, University of North Carolina, Chapel Hill, NC 27514)

It is apparent that the mechanism underlying comodulation masking release (CMR) relies on the existence of temporal fluctuations in the masker envelope. This raises the question of how much fluctuation is necessary to facilitate CMR. The present experiment addresses this question by measuring psychometric functions relating signal threshold to depth of flanker band modulation using sinusoidally amplitude-modulated pure tones as masker components. In the first set of conditions, the on-frequency component had a constant modulation depth of 100%, while the depth of the flanking components was varied between 100% and 0%. As expected, signal threshold improved monotonically as depth of flanker band modulation increased. In the second set of conditions, the on-frequency component had a constant modulation depth of 63%, while the depth of the flanking components was either 100%, 63%, or 0%. Results to date suggest that signal threshold is lowest when the flanking compo-

from 1 dB in the burst condition to 7 dB for a 450-ms fringe, due to a greater improvement in detectability in the correlated condition compared to the uncorrelated and all-uncorrelated conditions. When the cue bands were gated before the masker, the average CMR was larger (4–6 dB) than when the masker was gated before the cue bands (2–4 dB). These differences in improvement with increasing fringe duration may be attributable to differences in neural adaptation in the correlated and uncorrelated conditions. [Work supported by NINCDS Grant NS15895.]

2:30

**N5. Comodulation masking release (CMR) as a function of masker bandwidth, signal duration, and modulator bandwidth.** Gregory P. Schooneveldt and Brian C. J. Moore (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

Thresholds were measured for a 2000-Hz signal masked by continuous noise varying in bandwidth from 50 to 3200 Hz in 1-oct steps. For random noise maskers, thresholds increased with increasing bandwidth up to 400 Hz and then remained approximately constant. When the masker was amplitude modulated by a low-pass noise, so as to produce coherent envelope fluctuations across frequency, thresholds decreased as the masker bandwidth was increased beyond 200 Hz, giving a CMR. For a 400-ms signal duration, the CMR for masker bandwidths greater than 400 Hz increased from 2.4 to 12.3 dB as the modulator bandwidth was decreased from 400 to 12.5 Hz in 1-oct steps. For modulator bandwidths of 50 Hz or less, a release from masking of 3.5 to 7.3 dB occurred even for maskers with bandwidths of 50 and 100 Hz, less than the critical bandwidth at 2000 Hz. For a modulator bandwidth of 12.5 Hz, the CMR decreased from 12.3 to 5.3 dB as the signal duration was decreased from 400 to 25 ms in 1-oct steps. When the signal duration was less than 100 ms, there was no release from masking for masker bandwidths less than 400 Hz. The results suggest that, for maskers with fluctuating envelopes, across-channel comparisons can reduce signal thresholds even for short signals, but an extra within-channel process can produce a release from masking for long signals. This second process may reflect the ability of subjects to detect a change in the statistical properties of the envelope of the stimulus when the signal is added to the masker.

2:45–3:00

Break

nents have the same depth of modulation as the on-frequency component. [Research supported by AFOSR.]

3:15

**N7. Gap detection in a narrow-band noise with either a comodulated or a noncomodulated flanking band.** John H. Grose and Joseph W. Hall, III (Division of Otolaryngology Head and Neck Surgery, School of Medicine, University of North Carolina, Chapel Hill, NC 27514)

Comodulation masking release (CMR) suggests that the auditory system is sensitive to across-frequency differences in modulation pattern. This raises the question of whether it is as sensitive to modulation differences due to the absence of activity (a silent interval) as it is to the presence of additional activity (a signal). If so, gap detection in a narrow-band noise would be expected to be better in the presence of a comodulating flanking band than in the presence of a noncomodulating flanking band. The present study was designed to test this hypothesis. Gap detection was measured in a 30-Hz-wide narrow-band noise centered at 1 kHz. A second 30-Hz band of noise, centered at either 500 Hz or 1.5 kHz, was then added

that was either comodulated or noncomodulated with the 1-kHz band. While gap detection deteriorated with the addition of a second noise band, it appeared to do so more for the noncomodulated case than for the comodulated case. [Research supported by AFOSR.]

3:30

**N8. Cross-channel effects in amplitude-modulation detection.** Stanley Sheft and William A. Yost (Parmly Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

Sensitivity to low-frequency sinusoidal amplitude modulation (AM) at a single-component frequency of an equal-amplitude tonal complex was investigated. All masker components of the complex were modulated with a fixed depth at the same frequency as the probe. The phase between probe and masker envelopes was varied across conditions. For a two-component complex, the change in performance as a function of frequency separation ( $\Delta f$ ) depended on the envelope phase relationship. With a 2-kHz probe carrier, the slopes of the AM threshold versus  $\Delta f$  functions are more gradual than would be predicted by envelope interaction in a single-frequency channel. Consistent with AM detection models incorporating broad predetection filtering, performance may be affected by detection of changes in the overall modulation pattern. When the probe and masker envelopes differ, adding masker components at the fringes of the two-component complex can lead to an improvement in probe AM detection. This result suggests that the detection of the modulation pattern of an individual component in a tonal complex may be aided by enhancing cross-channel differences in modulation.

3:45

**N9. Complex sound discrimination: Predictions of the EWAIF model.** L. L. Feth, L. J. Stover, and R. A. Gerren (Department of Speech-Language-Hearing, University of Kansas, Lawrence, KS 66045)

The envelope-weighted average of instantaneous frequency (EWAIF) model of auditory perception has been successfully applied to simple, two-component signals, simultaneously amplitude- and frequency-modulated tones, and even to the complex signals used in the early "profile analysis" work. The EWAIF model predicts performance from a calculation of the envelope-weighted average of the instantaneous frequency fluctuations inherent in almost every complex periodic sound. When the model is in error, it generally predicts better performance than our listeners achieve. A revised version of the EWAIF model incorporates a temporal processing window. The revised model was tested by comparing its predictions with listeners' performance in a frequency-change discrimination task. The experiment requires the listener to distinguish a smooth frequency glide from a discrete, multistep transition over the same trajectory. The listeners' ability to distinguish the glide from the multistep transition, in a 2IFC task, decreased to chance as the number of steps increased. The EWAIF model performance follows that of the listeners,

given the appropriate choice of temporal window parameters. [Work supported by AFOSR and NIH.]

4:00

**N10. Detecting amplitude modulation of sinusoidal carriers.** William A. Yost and Stanley Sheft (Parmly Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

Listeners were asked to detect sinusoidal amplitude modulation (SAM) of one carrier tone (the target carrier) in the presence of a second carrier tone (the masking carrier) that also had SAM. The depth of modulation required to detect the presence of SAM of the target carrier was measured as a function of the difference in the frequency of amplitude modulation for the two carriers (modulator frequencies ranged from 4 to 100 Hz), the frequency separation between the two carriers (200- to 3000-Hz separation), and the phase of the sinusoidal modulator of the masker carrier relative to the phase of the sinusoidal modulator of the target carrier. In half of the conditions, the frequency of the target carrier was greater than the frequency of the masker carrier, while, for the other half of the conditions, the frequency of the target carrier was less than that of the masker carrier. The data will be discussed in terms of the strategies employed by the auditory system in processing temporal modulation of complex sounds. The data indicate that the auditory system uses a wide-band mode for processing temporal modulation, such that there is a great deal of interaction across widely separate frequency channels. [Work supported by the NINCDS.]

4:15

**N11. Modulation masking patterns.** Sid P. Bacon, D. Wesley Grantham, and Luann E. Van Campen (Division of Hearing and Speech Sciences, Vanderbilt University School of Medicine, Nashville, TN 37232)

Modulation thresholds for sinusoidally amplitude-modulated (SAM) broadband noise were obtained for modulation frequencies from 2 to 512 Hz using a two-interval, forced-choice adaptive procedure. The noise carrier was on continuously throughout a block of trials, and was modulated for 500 ms in one of the two observation intervals. Thresholds were obtained in quiet and in the presence of a SAM broadband noise masker. In the masking conditions, the same noise carrier, presented at a spectrum level of 15 dB SPL, was used for the signal and the masker. The masker was modulated in both of the 500-ms observation intervals. The modulation frequency of the masker was 4, 16, or 64 Hz; its modulation depth ( $m$ ) was 0 (no modulation), 0.5, or 1.0. For a given masker modulation frequency, the modulation masking patterns generally were bandpass, with the greatest amount of masking occurring when the signal and masker modulation frequencies were the same. With a few consistent exceptions, there was a monotonic relation between masker modulation depth and amount of masking (the greater the modulation depth the greater the amount of masking). These data indicate a tuning of the auditory system for the detection of modulation. [Work supported by NIH.]



**Session O. Underwater Acoustics II: Stochastic Volume and Boundary Scattering II**

John McCoy, Chairman  
Catholic University, Washington, DC 20064

Chairman's Introduction—1:30

*Invited Papers*

1:35

**O1. Imaging through random media.** Mark J. Beran (School of Engineering, Tel Aviv University, Ramat Aviv, Israel)

The distorting effects of random media on imaging are discussed. Random phase and random intensity fluctuations will be treated and the importance of isoplanicity will be emphasized. Methods of processing to overcome the distortion in conventional imaging will be considered, and the advantages of speckle interferometry and intensity interferometry will be mentioned.

2:00

**O2. Rough interface scattering.** David H. Berman, Dalcio K. Dacol, and John S. Perkins (U.S. Naval Research Laboratory, Code 5160, Washington DC 20375)

Recently, there has been a great deal of work on scattering by rough surfaces. Most of this work is aimed at describing scattering from the sea surface, and so Dirichlet boundary conditions are used. However, there are situations in which energy transmission across a boundary is important, such as in scattering from the seafloor. Hence, scattering by rough acoustically penetrable surfaces is considered. Methods for calculating such scattering will be briefly reviewed. Results obtained by the "point-matching" method will be presented for fluid-fluid and fluid-solid interfaces, both deterministic and random. Solutions of these problems yield forward and backscattering strengths of the seafloor, ice canopies, as well as the sea surface. In addition, it is possible to determine the probability distribution of scattered energy. These exact results will also be used to evaluate various approximation schemes.

2:25

**O3. Chaos in underwater acoustics.** F. D. Tappert, M. G. Brown (Applied Marine Physics, RSMAS, University of Miami, Miami, FL 33149), D. R. Palmer, and H. Bezdek (NOAA/Atlantic Oceanographic and Meteorological Laboratory, Miami, FL 33149)

The problem of predicting sound propagation in range-dependent ocean environments has been investigated, in which it is supposed that the environment (volume and/or boundary) varies smoothly in range and is exactly known with arbitrary precision. Although this problem as stated is deterministic and not intrinsically stochastic, it has been discovered from numerical and analytical studies of physically realistic examples drawn from deep ocean propagation, shallow-water propagation, and surface duct propagation, that ray path solutions exhibit "classical chaos," namely, unpredictable and stochastic behavior. Ray paths are found to have a continuous spectrum characteristic of noisy stochastic processes, and ray paths are found to have an exponentially sensitive dependence on initial conditions and environmental parameters characteristic of chaotic processes. This phenomenon of chaos in underwater acoustics is caused by the exponential proliferation of catastrophes (caustics) due to the loss of control implied by the nonseparability of variables in the eikonal equation. As a consequence, even when the ocean environment is known exactly, there exists a "predictability horizon" that limits the range to which acoustic fields can be predicted.

*Contributed Papers*

2:50

**O4. The limits of source localization due to stochastic ocean processes.** Terry E. Ewart, Robert P. Porter, Juraj Dworski, and Stephen A. Reynolds (Applied Physics Laboratory, University of Washington, Seattle, WA 98105)

The location of a source from acoustic measurements may be estimated in several ways. A known source/receiver geometry is used to study the

limits of geometrical techniques. Pulses (2, 4, 8, and 16 kHz) traveling a 6-km path from source to receiver in the Beaufort Sea were received at a 150-m vertical array during the AIWEX acoustic transmission experiment, AATE. Pulse travel times estimated using the Bell-Ewart multipath separation algorithm provide single-path travel time accuracies of a few  $\mu$  seconds. Treating the source location as unknown allows for the study of the precision of various methods. Ray trace methods were implemented to backpropagate the travel time estimates to a source location

using the mean sound-speed profile. A simple source location estimator based on least-squares fitting of the measured to the predicted travel times provides a classical example of a global versus local optimization problem when the array angle is included in the minimization. Methods have been developed to circumvent the optimization problem and range errors of a few centimeters and depth errors of a few meters result. The methods, results showing the effects of the medium, and a discussion of further refinements are presented. [Work supported by ONR.]

3:05

**O5. Chaotic behavior of ray trajectories in a range-dependent ocean environment.** D. R. Palmer (NOAA/AOML, 4301 Rickenbacker Causeway, Miami, FL 33149), M. G. Brown, F. D. Tappert (RSMAS/AMP, University of Miami, Miami, FL 33144), and H. F. Bezdek (NOAA/AOML, 4301 Rickenbacker Causeway, Miami, FL 33149)

It has been demonstrated that ray trajectories propagating in a range-dependent ocean environment can exhibit chaotic behavior. A particularly simple sound-speed model was considered consisting of the Munk reference sound-speed profile to which is added a small range-dependent, deterministic perturbation having an harmonic dependence on range and decreasing exponentially with depth. Chaotic ray trajectories were identified from an examination of Poincaré sections and power spectra. The sensitivity of chaotic trajectories to initial conditions and the consequent implications for predictability were investigated by considering the evolution with range of a bundle of rays that initially occupy a very small region in phase space. The largest Lyapunov exponent was determined by considering the spreading of the bundle. Since the ray equations define a nonautonomous Hamiltonian system with one degree of freedom, our results can be understood in terms of recent contributions to the study of classical chaos.

3:20

**O6. Chaos in shallow-water propagation.** F. D. Tappert and M. G. Brown (Applied Marine Physics/RSMAS, University of Miami, Miami, FL 33149)

Sound propagation in downward refracting shallow-water environments is considered, in which it is supposed that the water depth varies smoothly in range and is exactly known with arbitrary precision, and the bottom is perfectly reflecting at shallow angles. After a few additional assumptions are used to simplify the physics, it was found that the equations for ray paths (RBR) undergoing multiple bottom reflections reduce to the "standard map" of chaos theory, where the "stochasticity parameter"  $K$  is four times the ratio of boundary curvature to ray curvature. Typical values of  $K$  greatly exceed the critical value,  $K_c \approx 0.97$ , above which "global chaos" is known to prevail. Thus ray paths in these deterministic environments typically exhibit chaotic behavior, namely, ray paths have an exponentially sensitive dependence on initial conditions.

3:35

**O7. Application of the finite element method to simulating VLF/ULF under-ice acoustic propagation and scattering.** Stanley A. Chin-Bing (Naval Ocean Research and Development Activity, Numerical Modeling Division, NSTL, MS 39529-5004) and Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148)

A full-wave range-dependent ocean acoustic propagation and scattering model based on the finite element method [J. E. Murphy and S. A. Chin-Bing, *J. Acoust. Soc. Am. Suppl.* 1 **82**, S61 (1987)] has been applied to a shallow-water under-ice scenario at very low to ultra-low frequencies (VLF/ULF) and at extreme close range. The model has the advantage that it includes the coupled effects of the full-wave range-dependent forward propagated and backscattered fields. (However, the model does not yet include the effects of shear waves.) The under-ice surface was simulated by a fractallike interface superimposed on an ice keel representation. At very low frequencies, the acoustic wave can penetrate the ice-water interface and undergo reflection at the air-ice interface. The reflected, refracted, and backscattered waves couple to produce interesting interfer-

ence effects. Numerical simulations will be presented that examine the placement of receivers for optimum signal detection at these low frequencies. [Work supported by ONR/NORDA.]

3:50

**O8. Numerical modeling of the scattered acoustic field from elastic ice.** J. Robert Fricke (MIT/WHOI Joint Program, MIT, Cambridge, MA 02139), Ralph A. Stephen (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), and Arthur B. Baggeroer (Departments of Ocean and Electrical Engineering, MIT, Cambridge, MA 02139)

A two-dimensional elastic finite difference modeling program written for marine geophysical applications [M. E. Dougherty and R. A. Stephen, *J. Acoust. Soc. Am.* **82**, 238-256 (1987)] has been modified to study the ice scattering problem. The modified code was checked by comparing modeled plane-wave reflection coefficients at various angles of incidence with the analytical solutions provided by Zoeppritz equations. These tests were calibrated for a flat interface with homogeneous half-spaces of water and ice and resulted in favorable comparisons. Subsequently, various ice models were run to observe the scattered acoustic field. These models include flat ice, flat ice with a single keel, and ice with randomly rough surfaces (top and bottom) derived from different process models (e.g., Gaussian and Poisson). With this program, controlled numerical experiments can be performed for single realizations of ice roughness including full elastic behavior of the ice which is usually neglected in scattering theory. Such a capability can be used as an analysis tool for evaluating scattering approximations (say, the Kirchhoff approximation). In addition, this modeling capability can be used as the synthesis step of a scattering inversion program where ice roughness parameters are to be estimated from a measurement of the scattered field. [Work supported by ONR Arctic Acoustics program.]

4:05

**O9. Images of waves through random media by numerical simulation.** Stanley M. Flatté and Jan Martin (University of California, Santa Cruz, CA 95064)

Using the SDSC/CRAY, an extended random medium has been modeled by a set of two-dimensional thin phase-changing screens with phase power spectral densities having various power-law indices and inner and outer scales. The strength of the index of refraction in the medium has been varied. At small strength, the propagation is unsaturated, and the random intensity patterns at the final screen are dominated by the effects of the small-scale parts of the medium. As the strength of the medium fluctuations is raised, the images of intensity at the final screen show striking patterns, with caustics forming and interference fringes near the caustics being visible. For even larger medium strength, regions of high intensity associated with many caustics clump into large-scale patterns that are related to the large-scale structure of the medium. Intensity spatial spectra and variance will be given, but it should be emphasized that the observed patterns are not well characterized by a simple, random-phase realization of a power spectrum. Some comparisons with experiments in laser-beam propagation through the atmosphere will be given.

4:20

**O10. Field and numerical volume scattering experiments.** Terry E. Ewart and Stephen A. Reynolds (Applied Physics Laboratory and School of Oceanography, University of Washington, Seattle, WA 98105)

Until recently, the history of comparing ocean acoustic volume scattering experiments with theory bore scant resemblance to the traditional scientific method. However, vastly improved agreement between predicted and observed intensity correlations is now seen. It is clear that two elements of the research have played vital roles in reaching the current understanding. The first is the development of theories that include the parametrizations and approximations appropriate to the ocean case. The second is the characterization of the deterministic and stochastic processes of the ocean that give rise to the observed fluctuations. An overview of those processes relevant to the study of ocean volume scattering is presented, followed by a discussion of where attention needs to be focused

in order to expand our understanding of volume scattering. Results from the 1971 experiment at Cobb Seamount, the 1977 Mid-Ocean Acoustic Transmission Experiment, the 1985 AIWEX Acoustic Transmission Experiment, and numerical experiments provide the framework for the discussions. [Work supported by ONR.]

4:35

**O11. Acoustic detection in the weak scattering region of an oceanic internal wave field.** Bruce J. Bates (Naval Underwater Systems Center, Newport Laboratory, Newport, RI 02841)

The coherent pressure field of an acoustic harmonic point source is gradually transformed into an incoherent pressure field as a function of propagation distance through the oceanic internal wave field. A receiver located in the weak scattering region would measure a fluctuating acoustic pressure with a Gaussianly distributed logarithmic amplitude and phase. The consequence is a biased estimate for the conventional maximum likelihood estimate of pressure or intensity and subsequently a biased detection criteria. It is proposed that detection performance may be improved in this case by logarithmically transforming the pressure prior to detection processing. An analysis and discussion will be presented using the Garrett and Munk internal wave field model. [Work supported by NUSC IR/IED.]

TUESDAY AFTERNOON, 17 MAY 1988

WEST BALLROOM A, 1:45 TO 4:50 P.M.

### Session P. Physical Acoustics II: Martin Greenspan Memorial Session

Steven L. Garrett, Chairman

*Department of Physics, Code 61GX, Naval Postgraduate School, Monterey, California 93943*

Chairman's Introduction—1:45

#### *Invited Papers*

1:55

**P1. Martin Greenspan's work in acoustical physics.** Richard K. Cook (4111 Bel Pre Road, Rockville, MD 20853)

Martin (Moe) Greenspan's long career at the National Bureau of Standards began in 1936 in the general area of the strength of engineering materials and structures. His theory of stress distribution in perforated plates was an important contribution to modern linear elastic fracture mechanics. In 1947 he began to work in physical acoustics and continued in this until 1987. His landmark accomplishments include the theoretical analysis and measurement of sound propagation in rarefied gases, development of an instrument for measurement of sound velocity in sea water, and determination of the important physical influences leading to cavitation in liquids by ultrasound. In recent years, he worked on methods for measuring accurately the acoustical emissions (stress waves) generated by physically stressed materials of engineering. The influences of his work on sound in rarefied gases persist today; the database he obtained in the late 1940s–1950s was far in advance of the analytical capabilities for sound propagation in rarefied gases at that time. The data are still used for evaluation of new theoretical developments. Moe's work has always been characterized by an excellent coordination between theoretical–mathematical development and laboratory experiment.

2:20

**P2. Acoustic goofs<sup>a1</sup> or irreproducible effects in acoustics.** I. Rudnick (Physics Department, UCLA, Los Angeles, CA 90024)

Our respected and beloved friend, Moe, had a genius for being able to put his finger on mistakes that made their way into experiments and theories. He was a bubble buster. Listed below are some acoustic goofs. Moe contributed his skill in correcting some of them. There will be brief discussions of some of the topics as time allows: (1) Fitzgerald's folly; (2) The universal sequence in subharmonic generation or when is a decibel not a decibel?; (3) What is zero sound in liquid helium 4?; (4) Isothermal and adiabatic propagation in metals; (5) Shearing in longitudinal wave propagation; (6) Weber's folly. <sup>a1</sup> Webster's dictionary: goof—to make a mistake or blunder.

2:45

**P3. The dispersion of sound in monoatomic gases.** George W. Ford (Department of Physics, University of Michigan, Ann Arbor, MI 48109-1120)

The classical Kirchhoff derivation of the absorption of sound in gases at moderate pressures starts from the Navier–Stokes equations of hydrodynamics. At low pressures, specifically when the mean free path is no longer

small compared with the sound wavelength, this same derivation predicts a dispersion of the speed of sound. The absorption and dispersion can also be derived directly from the Boltzmann equation, with a result that confirms the Kirchhoff absorption coefficient, but with predicted dispersion and higher-order terms in the absorption that differ from those of hydrodynamics. Martin Greenspan's early work on sound propagation at low pressures provided the experimental data for a definitive test of the Boltzmann equation. These derivations will be briefly reviewed and the comparison with Greenspan's results will be shown. Some concluding remarks about the history of this problem and about more recent developments will be made.

3:10

**P4. Adiabatic invariance, the cornerstone of modern physics.** Seth J. Putterman (Department of Physics, UCLA, Los Angeles, CA 90024)

The article by M. Greenspan on the Boltzmann–Ehrenfest adiabatic principle [*J. Acoust. Soc. Am.* **27**, 34 (1955)] is used as the starting point for discussing the role of adiabatic invariance (or integrability) in modern physics. These conserved properties of open systems constitute the key concept upon which all advances in modern physics have been focused. These include the origin of thermodynamic irreversibility for Hamiltonian systems and the quantization of motion. Delauney (1860) was first to employ adiabatic invariants as dynamical variables with an aim toward determining the stability of the solar system. This issue eventually culminated in the work by Kolmogorov, Arnold, and Moser on the stability of nearly integrable systems. The solution to the problem of soliton motion by the inverse scattering transformation is in fact a transformation to the adiabatic invariant as the canonical momentum. Recent acoustical applications of adiabatic invariance include a determination of the potential in an acoustic levitator and a prediction of second sound in acoustic turbulence.

3:35

**P5. Gas-filled spherical resonators: The current state of experiment and theory.** James B. Mehl (Physics Department, University of Delaware, Newark, DE 19716) and Michael R. Moldover (Thermophysics Division, Center for Chemical Engineering, National Bureau of Standards, Gaithersburg, MD 20899)

In the past decade, spherical acoustic resonators have proved to be useful tools for high-accuracy measurements of the speed of sound in gases. Applications include thermophysical property determinations, acoustic thermometry, and the recent redetermination of the gas constant. The complex resonance frequencies of the resonator are described by a theoretical model that includes the effects of the viscous and thermal boundary layers, shell motion, imperfect spherical geometry, and other, less important, effects. The theoretical model is tested by comparing the consistency of speed-of-sound determinations using different modes, and by comparing the resonance half-widths with the theoretical predictions. The current level of agreement of experiment and theory will be reviewed.

4:00

**P6. Cavitation nuclei and cavitation noise.** M. Strasberg (David Taylor Research Center, Bethesda, MD 20884-5000)

Present understanding of the influence of nuclei on the onset of cavitation and the generation of radiated cavitation noise is reviewed. It is now generally accepted that the formation of cavities in a liquid at only moderate negative pressures requires the presence of nuclei in the liquid; the experiments of Greenspan and Tschiegg, among others, leave no doubt about this. However, there is still no agreement concerning the form of these nuclei or the mechanism causing them to persist under conditions that should ordinarily make them disappear. Various possible stabilizing mechanisms are discussed. It is shown that although nuclei determine whether cavitation occurs at all, once it occurs, their presence has little effect on the characteristics of the radiated noise generated by the cavitation. The factors that do influence the spectrum and the magnitude of the noise will be discussed, mainly in terms of hydraulically induced cavitation.

4:25

**P7. Theory of radiation induced cavitation.** Robert E. Apfel and Y.-C. Lo (Yale University, P. O. Box 2159, New Haven, CT 06520)

Data from two different experiments fall on the same general nondimensional curve: The first set of data is that of Greenspan and Tschiegg on neutron-induced acoustic cavitation [*J. Acoust. Soc. Am.* **72**, 1327 (1982)]; the second is that of Apfel *et al.* on neutron-induced (nonacoustic) cavitation of superheated drops [*Phys. Rev. A* **31**, 3194 (1985)]. The physics is the same. The theory begins with the neutron–nucleus interaction, continues as the recoil ion deposits its energy as heat along a line, and ends with the formation of a bubble that must exceed a critical size if a macroscopic bubble is to be produced. The theory will be reviewed and its predictions compared with both sets of experimental results. [Work supported by D. O. E. grant DE-FG02-87ER6050.]

**Session Q. Education in Acoustics II: *In-situ* Technical Tour of Northwest Acoustical Organizations**

Gerald F. Denny, Chairman

*Honeywell Marine Systems Division, 6500 Harbour Heights Parkway, Everett, Washington 98204*

Various Northwest organizations will have displays and representatives available from 8:00 a.m. to 12:30 p.m. to acquaint visitors with their facilities and their work in acoustics.

**Q1. Applied Physics Laboratory** (University of Washington, 1013 N.E. 40th Street, Seattle WA 98105). The laboratory focuses on education, research, and applied technology in ocean, polar, and environmental sciences.

**Q2. Boeing** (Seattle, WA 98124). The Boeing Commercial Airplane Noise Technology Staff is involved in the design and development of noise suppression hardware and noise prediction and testing technologies.

**Q3. David Taylor Research Center, Detachment Puget Sound** (Bremerton, WA 98314). The detachment conducts research, development, testing, and evaluation in underwater acoustics, and submarine noise reduction.

**Q4. Honeywell, Marine Systems Division** (6500 Harbour Heights Parkway, Everett, WA 98204). MSD develops and produces underwater transducers, sonar systems, and acoustic signal processing equipment.

**Q5. Naval Undersea Warfare Engineering Station** (Keyport, WA 98345). NUWES is active in research and measurement of underwater acoustic phenomenon.

**Q6. University of Washington** (Seattle, WA 98195). Departments that teach and carry out research in acoustics include Aeronautics and Astronautics, Bioengineering, Electrical Engineering, Mechanical Engineering, Music, Otolaryngology, Physics, and Speech and Hearing Sciences.

**Q7. Veterans Administration Medical Center** (1660 S. Columbia Way, Seattle WA 98108). The Audiology Clinic of the VA is carrying out cochlear implants as part of a research project for veterans with severe hearing losses. A demonstration of the signal processing strategies of three cochlear implants used in their cochlear implant project will be given.

**Session R. Engineering Acoustics III and Noise IV: Measurement of Pressure Fluctuations in Turbulent Boundary Layers**

Allan Zuckerwar, Chairman

*NASA Langley Research Center, M/S 238, Hampton, Virginia 23665*

Chairman's Introduction—8:00

*Invited Papers*

8:05

**R1. Acoustic sources in the turbulent boundary layer.** J. C. Hardin (NASA Langley Research Center, MS-461, Hampton, VA 23665)

This paper will be concerned with sources of sound in the simplest case of a turbulent boundary layer over a flat plate. For the majority of the paper, the driving flow will be considered to be subsonic such that gross compressibility effects, such as shocks, do not occur. Standard terminology, governing equations, and characteristics of the turbulence in the boundary layer and the noise it produces will be described. The paper will then examine the fundamental instability of this flow at high Reynolds number and the transition of a laminar boundary layer to turbulence. Generation of Tollmien-Schlichting waves and the myriad large scale structures (hairpin vortices, streaks, spots, bursts, etc.) that have been observed in boundary layers will also be described.

Finally, the theoretical understanding of noise production by this flow, which was initiated by Lighthill, Curle, and Powell and carried on by later research workers, will be described. In particular, the use of different Green's functions, which allows the noise to be determined either through integration of various flow quantities over the volume of the boundary-layer flow or by integration of the pressure over the flat plate, will be developed and the advantages of each noted. In light of this understanding, mechanisms for boundary-layer noise production and methods for their calculation are discussed. Additional sources in high-speed boundary layers with shocks present will also be described.

8:35

**R2. A semiempirical model for the wave vector-frequency spectrum of turbulent wall pressure on a smooth planar boundary.** D. M. Chase (Chase, Inc., Suite 510, 87 Summer Street, Boston, MA 02110)

The analytical framework and construction of a model of turbulent boundary-layer pressure on a planar wall by its relation to velocity-product fluctuations regarded as its sources are reviewed. Expansion for low (subconvective) wavenumbers and low Mach numbers provides a basis for such a trial model of the wall-pressure spectrum that is potentially satisfactory at all wavenumbers from the acoustic domain to the convective. The proposed form for source cross spectra incorporates the principle of wall similarity, a kinematic assumption about space-time correlation, possible wall-normal profiles of vorticity cross spectra, and assumed non-negativity. The wall-pressure model is obtained by analytical integration over the source-spectral profiles in limiting domains followed by convenient rough interpolation. The experimental basis for determination of parameter values of the model is considered by reference to classical wind-tunnel measurements dominated by the convective wavenumber domain and to more ambiguous ones directed to the subconvective but superacoustic domain. The inadequate state of validation and determination of parameters characterizing the acoustic domain and the domain of pertinence of the Kraichnan-Phillips theorem is confronted. Likewise, the open but experimentally resolvable question of the joint dependence on wave vector components in the convective domain is recalled.

9:05

**R3. The influence of surface roughness on wall-pressure fluctuations and the radiated sound from a turbulent boundary layer.** M. S. Howe (BBN Laboratories Inc., 10 Moulton Street, Cambridge, MA 02238)

The noise and vibration produced during turbulent boundary-layer flow over a nominally plane flexible surface are governed by the low wavenumber and acoustic regions of the wall-pressure wavenumber-frequency spectrum. In the case of a rough surface, the spectrum differs from that on a smooth wall on two counts: (i) the strengths of the turbulent Reynolds stress fluctuations, which are ultimately responsible for the pressure field, are increased by the action of surface roughness; (ii) the pressure field produced by those enhanced pressure sources is redistributed in the wavenumber plane by diffraction by the roughness elements. The wall-pressure spectrum can be expressed in the form  $P(\mathbf{k}, \omega) = P_0(\mathbf{k}, \omega) + P_R(\mathbf{k}, \omega)$ , where  $P_0$  denotes the spectrum that would be associated with the roughness-enhanced Reynolds stresses if the wall were assumed to be perfectly flat, and  $P_R$  is the additional component due to the diffraction mechanism (ii). It will be shown that  $P_R$  is expected to dominate the behavior of the wall pressure in the low-wavenumber and acoustic regions over a wide range of frequencies. Certain problems associated with experimental studies of rough wall boundary-layer noise will also be discussed.

9:35

**R4. Use of piezoelectric foil for measurement of pressure fluctuations and local shear stress in flight.** A. Bertelrud (High Technology Corporation, Hampton, VA 23366 and FFA, The Aeronautical Research Institute of Sweden, Bromma, Sweden)

The measurement of pressure fluctuations in air flows poses several serious problems to the experimentalist. Generally a very limited frequency response is available, as the cavity above ordinary pressure transducers plays a crucial role for the sensor response. This type of transducer requires extensive work for incorporation into a surface. Recently, piezoelectric foil has become available for various applications. It offers ease of application and distributed measurements without strict frequency response limits. However, although the data acquisition is straightforward, it appears that particular caution is required for interpretation of the data as pressure fluctuations. In the present paper, emphasis is put on frequency response, true root-mean-square determination, as well as interpretation into skin friction data. Through analysis of flight data and laboratory calibrations, it is shown that the piezoelectric foil in general will respond to pressure fluctuations, and it is also shown that the piezofoil constitutes a repeatable and rugged source of information.

10:05-10:20

Break

10:20

**R5. Bubble path in the wake of a cavitating propeller.** Mauro Pierucci (Department of Aerospace Engineering and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

Propellers that operate underwater at high rpm's cavitate at the tip. The tip cavitation creates air bubbles that are then swept downstream by the motion of the surrounding fluid. In this paper, a theory is presented to predict the local velocity and the path of the bubble. The bubble motion is assumed to be governed by a group of terms due to the acceleration of the displaced fluid, the convective term, and the drag due to the cross-sectional area of the bubble. At very low and very high Reynolds numbers, the equations have been solved in closed form. Results are presented for the bubble velocity and path for the following flow fields: (a) uniform axial flow field and (b) uniform flow field with an axially decaying swirling component. In all cases presented the bubble axial velocity component asymptotes to the free stream velocity; the manner in which it asymptotes is exponential at very low Reynolds numbers and algebraic at high Reynolds numbers. Bubble helical paths and velocity patterns are shown for different bubble sizes.

10:35

**R6. Extended frequency response of a fiber-optic microphone.** Allan J. Zuckerwar (NASA Langley Research Center, MS-238, Hampton, VA 23665) and Mathew L. Areford (Hampden-Sydney College, Hampden-Sydney, VA 23943)

A condenser microphone responds to the mean displacement of a stretched membrane excited into an axially symmetric mode of vibration. At some frequency between the first and second resonant frequencies, the mean displacement falls to zero and the microphone response drops sharply after the first resonance. A fiber-optic microphone, on the other hand, responds to the displacement at the center of the membrane. The center displacement of an axially symmetric mode never falls to zero. Consequently, through judicious backplate design, the membrane motion can be damped to achieve a smooth merger in response between the first and second resonances, and the bandwidth of the microphone can be extended to beyond the second resonant frequency. After a brief review of the construction of the condenser and fiber-optic microphones, the results of a theoretical analysis will be presented for two cases: (1) a B&K  $\frac{1}{8}$ -in. condenser microphone compared with its fiber-optic equivalent, and (2)  $\frac{1}{8}$ -in. fiber-optic microphone of special design to measure pressure fluctuations in a turbulent boundary layer.

10:50

**R7. A callibrator for the measurement of the static and dynamic shear stresses along a stretched Mylar membrane.** Frank W. Cuomo (Department of Physics, The University of Rhode Island, Kingston, RI 02881-0817 and Naval Underwater Systems Center, Newport, RI 02841-5047)

Recent power spectral density measurements of the pressure fluctuations in a turbulent boundary layer obtained with a fiber-optic lever pressure sensor in a low-speed wind tunnel have compared favorably with data reported in the literature. Utilizing a similar sensor arrangement, incorporating a flush-mounted, metallized Mylar membrane, the measurement of shear stresses leading to skin friction has been contemplated. The first step toward the realization of a shear sensor has been to establish the behavior of a stretched Mylar membrane acted upon by forces parallel to the flow direction. A callibrator designed to meet the requirements normally found in a wind-tunnel environment has been constructed and tested. In this paper, its operation is described and data of Mylar samples are presented. The resolution of the device for static measurements is based on translations on the order of  $1 \mu\text{in.}$  while the dynamic response covers a frequency range of dc to 18 kHz.

11:05

**R8. Comparison between design and installed acoustic characteristics of the NASA Lewis 9- $\times$ 15-ft low-speed wind-tunnel acoustic treatment.** Milo D. Dahl and Richard P. Woodward (National Aeronautics and Space Administration, Lewis Research Center, Cleveland, OH 44135)

The test section of the NASA Lewis 9- $\times$ 15-ft low-speed wind tunnel was acoustically treated to allow the measurement of sound under simulated free-field conditions. The treatment was designed for high sound absorption at frequencies above 250 Hz and to withstand the environmental conditions that exist in the test section. To achieve the design requirements, a fibrous bulk absorbing material was packed into removable panel sections. Each section was divided into two equal-depth layers that were packed with material to different bulk densities. The lower density was next to the face of the treatment. The facing consisted of a perforated plate and screening material layered together. Sample tests for normal incidence acoustic absorption were conducted in a small rectangular duct. Tests with no air flow involving the measurement of the absorptive properties of the installed treatment combined the use of time delay spectrometry with a previously established free field measurement method. These measurements were compared to the design model and sample tests of the treatment.

**Session S. Physical Acoustics III: Lord Rayleigh's Contributions to Acoustics**

Honorable Guy R. Strutt, Cochairman  
*Old Rectory, Terling, Chelmsford, Essex CM3 2QE, England*

Malcolm J. Crocker, Cochairman  
*Department of Mechanical Engineering, Auburn University, Auburn, Alabama 36849*

Chairman's Introduction—8:10

*Invited Papers*

8:15

**S1. Lord Rayleigh and his influence on science and scientists.** Malcolm J. Crocker (Department of Mechanical Engineering, Auburn University, Auburn, AL 36849)

John William Strutt, the third Lord Rayleigh (1842–1919), was a man of considerable scientific accomplishments. He contributed to theoretical and experimental physics and was one of the great pioneers in scientific and engineering methods. He was also a man who moved in the highest social and intellectual circles of his day. He was a relative of prime ministers and received almost every honor to which a British scientist could aspire. He was a Nobel prize winner and President of the Royal Society. His book, *The Theory of Sound*, has remained in print for over a century. From 1880 to 1885, after J. C. Maxwell's death, Lord Rayleigh directed the famed Cavendish Laboratory at Cambridge University, but after that mainly lived in Terling Place, his country house near London where he established a remarkable scientific laboratory in which he later discovered argon. During his lifetime Lord Rayleigh published over 400 scientific papers and carried on a vigorous correspondence with scientists such as Maxwell, Kelvin, Stokes and others. This paper will examine his papers and correspondence to put his scientific contributions, particularly in acoustics, into historical perspective.

8:45

**S2. Rayleigh's philosophy of problem solving, as inferred from his original papers on acoustics.** Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

Rayleigh not only had a knack for skimming off the cream, but also for finding cream where others might have thought there was no cream to be found. His first substantial work of note, the 1870 memoir on resonance in the *Philosophical Transactions*, introduced a style and philosophy that is evident in many of his later works and that is worth emulating. In this paper, Rayleigh successfully bypassed the elegant but difficult to apply mathematics of Helmholtz's epochal 1860 paper, and substituted instead a few simple approximate concepts and produced some relatively simple formulas that could be compared with Sondhauss's experimental results on what we now call Helmholtz resonators. Other examples cited in the present paper are Rayleigh's theory of diffraction by orifices and slits, his analytical models for elastic shells, his discovery of adiabatic invariants in his 1902 "pressure of vibrations" paper, and his approach to modeling low-frequency sound radiation from sources. Rayleigh's work and philosophy may have had a substantial influence, although perhaps not consciously recognized, on modern trends such as finite elements and matched asymptotic expansions. In addition to a thorough familiarity with basic physical concepts, a good intuitive feel for what was relevant, and an inherent love of simplicity, Rayleigh made extensive use of variational techniques as a framework for developing approximations. Hitherto they had been used primarily as a means for elegant expression of basic principles. Also noteworthy was Rayleigh's heavy use of similitude arguments. [Work supported by ONR.]

9:15

**S3. Rayleigh's influence on engineering vibration theory.** Stephen H. Crandall (Massachusetts Institute of Technology, Cambridge, MA 02139)

The first ten chapters of *The Theory of Sound* constitute an unprecedented exposition of vibration theory. This tour de force set the mold for the teaching of vibration for more than a century. The central core is linear modal analysis applied to lumped parameter systems and to strings, bars, beams, membranes, plates, and shells. In addition, there are preliminary investigations of nonlinear vibrations, parametrically excited vibrations, and random vibrations. Most of the textbooks on vibrations for engineers that have appeared in the past 50 years may be viewed as attempts to explain Rayleigh's idea to university students. Rayleigh's paradigm has served us



well for a century but it appears that we are now on the threshold of major changes in the way vibration theory is taught. These changes are being driven by the tremendous advances in instrumentation and data processing facilities. The second coming of modal analysis (modal testing, really) is largely based on the concept of transfer functions. Although Rayleigh occasionally used the complex exponential and even computed a few transfer functions in passing, he did not attach any special significance thereto. It is interesting to speculate what Rayleigh would have done had he envisioned today's multichannel spectral analyzers. Other advances, unavailable to Rayleigh, which are currently entering vibration courses, are discussed.

9:45

**S4. Lord Rayleigh and reciprocity in physics.** Richard K. Cook (4111 Bel Pre Road, Rockville, MD 20853)

In *The Theory of Sound*, Rayleigh presents his analysis of the reciprocal theorem of mathematical physics with the observation that "Very remarkable reciprocal relations exist between the forces and motions of different types... ." in a linearized dynamical system. He introduces the reciprocity principle for sound waves in an air-filled space: "On his extension of Green's theorem, Helmholtz founds his proof of the important (reciprocal) theorem contained in the following statement: ... ." In the ensuing 100 years since publication of his book, the reciprocity principle has been extended to many areas of physics—electrodynamics, electroacoustics, piezoelectricity, electromagnetic waves, optics, etc. The principle has been successfully applied in electroacoustics to achieve a system for accurate, absolute calibration of microphones used for acoustical measurements. The system is now internationally used by standards laboratories. Limitations and restrictions on applicability of the reciprocity principle to certain physical systems have come to light. Conspicuous examples involve the propagation of sound in moving air and the propagation of electromagnetic waves through magnetized media.

10:15

**S5. Lord Rayleigh's contributions to musical acoustics.** Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

Rayleigh's work touched virtually every area of acoustics in some way; musical acoustics is certainly no exception. Besides laying the groundwork for all subsequent research on musical instruments with his informative discussions of the vibrations of strings, bars, membranes, plates, air columns, and resonators, he did important theoretical and experimental work on bells and kettledrums. He identified the vibrational modes that radiated the most prominent partials in the sounds of both these instruments, and he recognized that the nominal pitch of a church bell lies one octave below the fifth partial. In the area of music perception, his most important contributions were his studies of absolute pitch and sound source localization.

10:45

**S6. Lord Rayleigh's contributions to aeroacoustics.** Alan Powell (Department of Mechanical Engineering, University of Houston, Houston, TX 77004)

The growth of aeroacoustics as a distinct discipline burgeoned at midcentury, initially fostered by the need to control jet noise. It merged the then separate disciplines of aerodynamics and sound. Rayleigh knew no such boundaries and would likely have made major contributions to aeroacoustics if the fundamental questions had captured his attention. As it was, he wrote on many highly pertinent aspects: e.g., his theory for sensitive flames and jets (visual sound indicators) led to the modern instability theory of jets and other important aerodynamic flows, while the hypothesized feedback in his "bird-call" (a compatible high-frequency sound source) is similar to that in edge tones and in many other flow resonances. These had puzzling inconsistencies until Rayleigh's method of similitude was applied through Reynolds and Strouhal numbers. He had applied it to Strouhal's aeolian tones and to the scattering of sound from inhomogeneities, in which he introduced the "acoustic model" of aerodynamic sound theory (and in which the method for Rayleigh's integral was used for plane boundaries). There are now hundreds of research publications in aeroacoustics, some notable books and a place in university education: Rayleigh's name occurs many times.

11:15

**S7. Lord Rayleigh's contributions to nonlinear acoustics.** Robert T. Beyer (Department of Physics, Brown University, Providence, RI 02912)

While the overwhelming majority of Rayleigh's contributions to acoustics have been in the linear domain, there are many nonlinear topics on which he made early progress and set the stage for further progress in the future. His work on radiation pressure, cavitation, shock waves, and flow-induced sound will be discussed and the developments that followed from them will be described.

11:45

**S8. Rayleigh's observations of wave interference, evanescent fields, and sound cancellation.** David C. Swanson (Textron Defense Systems, Acoustic/Seismic Sensor Technology Group, 2385 Revere Beach Parkway, Everett, MA 02149)

In Chap. 23 of *The Theory of Sound*, Rayleigh made some outstanding observations of wave interference in an experiment designed to show that the source of "grave notes" in pipe organs was due to difference tones rather than a nonlinearity in the ear as proposed by Helmholtz [J. W. S. Rayleigh, *The Theory of Sound* (Dover, New York, 1945), pp. 461-462].

When the organ keys for  $d'''$  (1174 Hz) and  $e'''$  (1318 Hz) are played simultaneously, a grave note of 144 Hz (between a C# and D) is also heard, giving a displeasing sound. Rayleigh carefully tuned the pipes so that the grave note gave the slowest possible beat with a tuning fork of 128 Hz. Rayleigh noted that if the grave note was due to the ear, the sound of the tuning fork would not cause the slow cycles of sound cancellation. He also noted that the sound field from the fork-pipe combination (during constructive interference) fell off more rapidly with distance than the sound field from the fork alone, which is characteristic of an evanescent or nearfield. These observations provide an interesting lesson for all experimentalists on basic scientific reasoning and the importance of reporting all relevant observations revealed by experiment.

WEDNESDAY MORNING, 18 MAY 1988

GRAND BALLROOM C, 8:30 TO 9:50 A.M.

### Session T. Speech Communication III: Neural Networks in Speech Recognition I

Les Atlas, Chairman

Department of Electrical Engineering, FT-10, University of Washington, Seattle, Washington 98195

Chairman's Introduction—8:30

#### Invited Papers

8:35

**T1. Neural nets for speech recognition.** William Y. Huang, Richard P. Lippmann, and Thao Nguyen (P. O. Box 73, MIT Lincoln Laboratory, Lexington, MA 02173)

Artificial neural networks are of interest for two main reasons. First, they provide architectures to implement many algorithms used in speech recognizers with fine grain massive parallelism. Second, they are leading to new computational algorithms and new approaches to speech recognition inspired by biological nervous systems. Neural net approaches to the problems of speech preprocessing, pattern classification, and time alignment are reviewed. Preprocessors using auditory nerve time-synchrony models have provided improved recognition performance in noise [O. Ghitza, ICASSP 87, 2372-2375]. Highly parallel neural net architectures exist to implement many important traditional classification algorithms, such as  $k$ -nearest neighbor and Gaussian classifiers [R. Lippmann, IEEE ASSP Mag. 4(2), 4-22 (1987)]. Newer multilayer perceptron classifiers trained with back propagation can form arbitrary decision regions, are robust, and train rapidly for convex decision regions. These nets performed as well as conventional classification algorithms for digit and vowel classification tasks [W. Huang *et al.*, 1st ICNN IV, 485 (1987)]. Neural net approaches to the problem of time alignment are reviewed, including a neural net model called a Viterbi net that implements the Viterbi decoder used successfully in continuous distribution hidden Markov models (HMMs) [R. Lippmann *et al.*, 1st ICNN IV, 417 (1987)]. [This work was sponsored by the Defense Advanced Research Projects Agency and the Department of the Air Force.]

9:00

**T2. Speech recognition using time-delay neural networks.** A. Waibel, T. Hanazawa, K. Shikano (ATR Interpreting Telephony Research Laboratories, Twin 21 Tower, Osaka 540, Japan), G. Hinton (University of Toronto, Toronto, Canada), and K. Lang (Carnegie-Mellon University, Pittsburgh, PA 15213)

A time-delay neural network (TDNN) approach is presented to speech recognition that is characterized by two important properties: (1) Using multilayer arrangements of simple computing units, a TDNN can represent arbitrary nonlinear classification decision surfaces that are learned automatically using error back propagation. (2) The time-delay arrangement enables the network to discover acoustic-phonetic features and the temporal relationships between them independent of position in time and, hence, not blurred by temporal shifts in the input. The TDNNs are compared with the currently most popular technique in speech recognition, hidden Markov models (HMM). Extensive performance evaluation shows that the TDNN recognizes voiced stops extracted from varying phonetic contexts at an error rate four times lower (1.5% vs 6.3%) than the best of our HMMs. To perform this task, the TDNN "invented" well-known acoustic-phonetic features (e.g.,  $F2$

rise, *F2* fall, vowel onset) as useful abstractions. It also developed alternate internal representations to link different acoustic realizations to the same concept. The TDNNs trained for other phonetic classes achieve similar high levels of performance. The integration of such smaller networks into large phonetic nets and propose strategies for the design of neural network based large vocabulary speech recognition systems is discussed.

9:25

**T3. Speech recognition with layered networks.** D. J. Burr (Bell Communications Research, Morristown, NJ 07960)

Experiments show that perceptrons may be useful for speech recognition. Perceptrons, or layered networks, can recognize spoken digits, polysyllabic words, and alphabetic text with accuracies exceeding 98%. They do so with varied input representations ranging from Fourier through cepstral and linear prediction coefficients. As in conventional classifiers, the greater the number of training tokens per class, the higher is the accuracy. Trained networks may be analyzed to observe any structure in the weight space. One can see evidence in perceptron networks for both *voicing* and *formant onset* detection capability. For a given problem, a layered network has a *critical* number of hidden units that maximizes recognition accuracy at minimum hardware cost. Additional hardware may be saved by eliminating near-zero weights in upper layers as these have minimal effect on recognition accuracy. A tradeoff may exist between number of weights and network complexity. A small layered network with 264 weights and 12 hidden units achieved 99% accuracy on isolated digits. In contrast, a large perceptron with more than 11 000 weights and *no* hidden units achieved 98.2% accuracy on the difficult *E-set* alphabet and perfect recognition on twenty polysyllabic words. Apparently, memory (number of weights) can be traded for logic (number of hidden neurons) as in conventional computer programs.

WEDNESDAY MORNING, 18 MAY 1988

EAST BALLROOM A, 8:30 A.M. TO 12:05 P.M.

### Session U. Underwater Acoustics III: Stochastic Volume and Boundary Scattering III

Eric I. Thorsos, Chairman

*Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105*

Chairman's Introduction—8:30

#### *Contributed Papers*

8:35

**U1. Reflections from a block of sea ice.** Gerald R. Garrison and Robert E. Francois (Applied Physics Laboratory, University of Washington, 1013 NE 40 Street, Seattle, WA 98105)

At an Arctic ice camp in the Beaufort Sea, monostatic acoustic reflections near normal incidence were measured from the lower face of an isolated block of natural sea ice. The measurements were repeated with the transition layer sawed off to leave a surface of solid ice. A comparison of the two results showed that the 15-cm-thick skeletal layer and transition zone cause a reduction in reflectivity, beyond that predicted from the impedance mismatch, that increases from 10 db at 20 kHz to 30 dB at 80 kHz. The system was calibrated using the air-water interface formed at the open bottom of a shallow hollow metal cylinder. Air was forced into the cylinder when it was in place to fill it and thus provided a flat pressure-release reflecting surface. As the measuring transducer was moved away from normal incidence, the response pattern was found to agree with the sidelobe pattern predicted for a flat, perfectly reflecting surface. These measurements of the effect of the skeletal layer on ice reflectivity will be helpful in modeling the reflections from ice keels. [Work supported by the Office of Naval Technology with technical management by NORDA.]

8:50

**U2. Reflectivity of sea ice.** Robert E. Francois, Gerald R. Garrison, and Timothy Wen (Applied Physics Laboratory, University of Washington, 1013 NE 40 Street, Seattle, WA 98105)

The reflections from the face of a finite block of sea ice are usually predicted as the combination of two phenomena, as allowed under the Kirchhoff approximation. One is the reflection loss due to the change in acoustic impedance at an infinite water-ice boundary. The other is the interference pattern produced by the coherent return from a finite block face. Measurements of the reflections from a block of natural sea ice, described in a companion paper, showed that the reflection is further reduced by the skeletal layer. More information on the acoustic properties of this layer is needed in order to model its effect. Reflection losses are described by an amplitude reflection coefficient which would be 0.34 due to the impedance mismatch at the water-ice boundary. Reflection coefficients measured for 1- to 2-m-thick ice in the Arctic at frequencies 20-80 kHz are compared with measurements by Stanton and Jezek for thin ice. During the Arctic measurements, the return from the back face of the block was also detected, providing information on the absorption of sound in the ice. Field experiments have been designed to separate the absorption

in the skeletal and transition layer from the absorption in solid ice. [Work supported by the Office of Navy Technology with technical management by NORDA.]

9:05

**U3. New findings on scattering from floating plate with lateral heterogeneities.** Jacques R. Chamuel (Sonoquest/Advanced Ultrasonics Research, P.O. Box 153, Wellesley Hills, MA 02181) and Gary H. Brooke (Defence Research Establishment Pacific, FMO Victoria, British Columbia V0S 1B0, Canada)

Experimental ultrasonic modeling results are presented on near grazing incidence scattering from cracks, ridges, and edges of floating plates bounding a shallow water waveguide. It was observed that underwater acoustic waves propagating in a shallow water waveguide bounded by a floating plate produce a ringing wave component in the water upon scattering from a crack or a plate edge. This ringing component is associated with the water wave, travels with the water wave velocity, and its time of arrival is independent of the length of plate region along the wave path. The ringing frequency, however, is determined by the shear wave velocity and the plate thickness. The plate thickness corresponded to half a wavelength based on the shear wave velocity. The contribution of the ringing wave component to scattering from wet cracks, dry cracks, and ridges are demonstrated. The impact of these new findings on interpreting laboratory scaled studies on Arctic acoustics are discussed in view of the wide use of Plexiglas to model sea ice. Examples are given on scattering from single and multiple cracks and ridges. [Work supported by DREP and ONR.]

9:20

**U4. An application of a full scattering theory to Arctic propagation.** J. S. Hanna, D. M. Rubenstein, L. B. Dozier, and C. Pearson (Science Applications International Corporation, 1710 Goodridge Drive, McLean, VA 22102)

The full scattering theory of Burke and Twersky [J. Acoust. Soc. Am. **40**, 883–895 (1966)], which includes both the specular and scattered energy for a rough surface consisting of elliptical bosses on a plane surface, is applied to the problem of propagation under the ice surface. This theory has been efficiently parametrized [D. Rubenstein (unpublished manuscript)] for the case of randomly spaced keels with a Rayleigh distribution of depths. The scattering kernel for this treatment has been used in a coupled mode formulation of the ASTRAL propagation model. The modified ASTRAL can compute the propagation associated with both the specular field and the total field which includes the nonspecular component. The redistribution of energy among the modes occurs continuously with range, so the model can display the mode excitation as a function of range. The model is compared to some measured propagation data, particularly that of Diachok [J. Acoust. Soc. Am. **59**, 1110–1120 (1976)], who applied low- and high-frequency asymptotic forms of the Burke-Twersky theory for just the specular field. [Work supported by the ASW Environmental Acoustic Support Program.]

9:35

**U5. High-frequency reflection and scattering by multicomponent rough surface distributions.** R. J. Lucas<sup>a)</sup> and V. Twersky (Mathematics Department, University of Illinois, Chicago, IL 60680)

Earlier forms for coherent reflection and incoherent scattering by mixtures of different bosses on rigid or free base planes [V. Twersky, J. Acoust. Soc. Am. **29**, 209–225 (1957)] are applied to recent results for triaxial ellipsoids [R. J. Lucas and V. Twersky, J. Acoust. Soc. Am. Suppl. **1** **81**, S20 (1987)] to investigate continuous distributions. Key variables  $d$  (e.g., keel depth) are governed by two-parameter probability density functions  $f(m, v; d)$  with  $m$  as the mean value of  $d$ , and  $v$  as the normalized variance whose value (zero to unity) determines the curve of  $f$ . For  $v$  small,  $f$  is Gaussian (and reduces to a delta function as  $v$  ap-

proaches zero to reproduce one-component results); more generally, the curve is skewed, and as  $v$  approaches unity,  $f$  reduces to the exponential for the simplest Poisson case. Numerical integrations over finite ranges of  $d$  provide illustrative graphs to exhibit effects of truncation and parameters on the reflection coefficients and differential scattering cross sections per unit area. [Work supported by ONR.] <sup>a)</sup> Visiting from Department of Mathematical Sciences, Loyola University, Chicago, IL.

9:50

**U6. Application of Twersky's boss scattering theory to laboratory measurements of sound scattered by a rough surface.** D. Chu and T. K. Stanton (1215 West Dayton Street, Geophysical and Polar Research Center, University of Wisconsin, Madison, WI 53706)

One of the many challenges in describing the scattering of sound by rough surfaces is to address the fact that most surfaces are three dimensional. Furthermore, only their statistical properties may be known. Twersky's model [V. Twersky, J. Acoust. Soc. Am. **29**, 209–225 (1957)], by using an image method and taking multiple scattering into account, established a general solution involving distributions of three-dimensional bosses. A prominent advantage of the theory over others is that it can describe bosses of arbitrary orientation. The purpose of the present paper is to compare the theory with laboratory measurements of sound scattered by a continuously rough pressure-release surface. In the model, we use prolate spheroidal bosses oriented in a range of directions and randomly packed to approximate the rough surface. The data were taken from Welton [P. J. Welton, report ARL-TR-75-30, University of Texas at Austin (1975)] and involve backscattered sound versus grazing angle for several frequencies. The comparisons of the numerical computations with Welton's data show a reasonable agreement with respect to not only the boss orientation parameters but also the surface statistical parameters (rms roughness and correlation length). An interesting phenomenon is that the number of spheroidal modes required to fit the data increases as the frequency of the incident plane wave increases.

10:05

**U7. Diffraction of harmonic acoustic radiation by the apex of a ridge.** Michael J. Buckingham (Radio and Navigation Department, Royal Aircraft Establishment, Farnborough, Hampshire GU14, 6TD, England)

In general, there are three components in the acoustic field propagating over a ridge: an arrival via a direct path, which does not interact with the ridge, a component diffracted by the apex of the ridge, and a component reflected from the boundary of the ridge. All three contributions are included in a new, exact solution of the Helmholtz equation for the three-dimensional, harmonic field from a point source over a ridge formed by two intersecting, pressure-release planes. The solution is obtained by the application of three integral transforms to the Helmholtz equation, followed by the corresponding inverse transforms. It is valid for any ridge angle  $\theta_0$ , including the special case of a knife edge ( $\theta_0 = 360^\circ$ ) and a plane ( $\theta_0 = 180^\circ$ ). In the latter case, which is the Lloyd's mirror problem, there is no diffraction; only the direct and reflected components contribute to the field, which is highly variable because of strong spatial interference. Diffraction by the knife edge is exemplified by the leakage of energy over the barrier, and by the presence of spatial interference in regions where there is no reflected component. Similar phenomena appear with other ridge geometries, where it is usually possible to assess whether interference effects are due predominantly to diffraction or reflection.

10:20

**U8. Acoustic fields scattered simulated three-dimensional wind-driven sea surfaces.** Jerald W. Caruthers (NAVOCEANO, Geophysics Department, NSTL, MS 35925), Richard S. Keiffer (NORDA, Numerical Modeling Div., NSTL, MS 35925), and Jorge C. Novarini (Syntek Engineering & Computer Systems, Inc., 6110 Executive Boulevard, Suite 320, Rockville, MD 20852)

Statistical characteristics of acoustic fields scattered in the forward direction from simulated three-dimensional wind-driven sea surfaces are studied for frequencies from 0.1–10 kHz, high grazing angles, and ranges in the Fresnel region. The simulated surfaces have the Pierson–Moskowitz directional spectrum for a 5-m/s wind speed. Complex pressures are calculated using Helmholtz–Kirchhoff theory with a minimum number of approximations. Statistics of the acoustic fields are calculated as ensemble averages of the complex pressures and their moments. Comparisons are presented for differences in the ensemble of surfaces, spectral forms, surface illumination patterns, and implementation of the scattering theory. The technique allows for the accurate representation of first- and higher-order moments of the acoustic field scattered from a three-dimensional surface. Such quantities are usually difficult to obtain by other techniques. [This work has been supported by NAVOCEANO and NORDA.]

10:35

**U9. The effect of finite wind fetch on the acoustic field scattered from wind-driven sea surfaces.** Jorge C. Novarini (Syntek Engineering & Computer Systems, Inc., 6110 Executive Boulevard, Suite 320, Rockville, MD 20852), Jerald W. Caruthers (NAVOCEANO, Geophysics Department, NSTL, MS 39529), and Richard S. Keiffer (NORDA, Numerical Modeling Division, Code 221, NSTL, MS 39529)

The effect of a finite fetch on the acoustic field scattered by the ocean wind-driven surface for various wind speeds is analyzed through numerical simulation. The JONSWAP spectrum is adopted to describe the surface statistics of growing seas and the Pierson–Moskowitz spectrum for fully developed seas. Ensembles of three-dimensional surfaces having the desired directional spectra are numerically generated. For each surface realization, the complex pressure of the forward scattered field is calculated within the framework of the Helmholtz–Kirchhoff theory for  $kr > 1$ . Statistical moments of the scattered field are then calculated as ensemble averages. The fields scattered in the windward and cross-wind directions are evaluated for frequencies up to 1.5 kHz and for angles away from grazing incidence. Results indicate that the scattered field is sensitive to the difference between the JONSWAP and the Pierson–Moskowitz spectra. In particular, the total intensity and the coherence of the scattered field are significantly affected by the wind fetch. [Work supported by NAVOCEANO and NORDA.]

10:50

**U10. Resonator systems for *in situ* bubble measurements near, and at, the sea surface.** Nigel D. Breitz<sup>a)</sup> and Herman Medwin (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

A parallel plate acoustical resonator has been developed to measure bubble densities and their fluctuations near the sea surface and under whitecaps. Calibration has been achieved with microbubbles whose size (radius approximately  $50 \mu$ ) was evaluated by speed of rise in an aquarium. Using the “sea” generated in a 20-m-long water–wind tunnel, the resonator performance has been shown to be independent of orientation, depth, and sea surface roughness. A second device, using an upward-looking resonator, is also being developed to investigate larger bubbles at the surface. *In situ* acoustical devices are compared with remote sensing techniques, and early results of ocean trials are revealed. [Work supported by ONR.] <sup>a)</sup> Ocean Acoustics Associates, 4021 Sunridge Road, Pebble Beach, CA 93953.

11:05

**U11. Acoustic phase measurements from volume backscatter.** R. Del Huston (Arctic Sciences Ltd., 1986 Mills Road, Sidney, British Columbia V8L 3S1, Canada)

Stable and useful measurements of the orientation of the acoustic signal vector or acoustic phase as a function of range and time can be ob-

tained from volume backscatter. Individual targets that are normally random and sparse relative to the acoustic wavelength, yield echoes with random phases between independent realizations. This difficulty has been overcome by complex superposition of echoes from successive transmissions, such that the effective density of acoustic targets progressively increases as the summation proceeds. The theoretical basis of this type of coherent processing has been developed and examined in the limiting case, in which it approximates a scatterer continuum. The theory provides fundamental insights to the behavior of both the amplitude and phase of composite echoes from a volume mirror. Experimental work in Saanich Inlet, B.C. serves to confirm the theoretical concepts through a rms difference measurement of the acoustic phase which removes fluctuations from the evolving sound-speed profile. Based on these results, the potential for new types of oceanographic measurements using these techniques appears both realistic and encouraging.

11:20

**U12. Investigation of 12-kHz-deep scattering layers observed with the multibeam echo-sounder Sea Beam.** Christian de Moustier (Marine Physical Laboratory, Scripps Institution of Oceanography, La Jolla, CA 92093) and Dimitri Alexandrou (Department of Electrical Engineering, Duke University, Durham, NC 27706)

During a survey over Horizon Guyot in March 1986, acoustic measurements in the deep scattering layers were carried out with the 12-kHz multibeam echo-sounder Sea Beam installed on the R.V. ATLANTIS II. Acoustic returns received by each of the Sea Beam’s 16 preformed beams were sampled in quadrature at 1 kHz from the surface down to about 1400-m depth, and recorded digitally with a special purpose data acquisition system originally developed at the Marine Physical Laboratory for seafloor acoustic backscatter measurements. These complex volume reverberation data are presented as horizontal and vertical cross sections showing the 3-D structure of the scattering layers observed. Statistical analyses of the data involve homogeneity and stationarity tests on a beam to beam level as well as on a single beam along the ship’s track to establish the degree of patchiness and the corresponding reverberation levels associated with the layers observed. [Work supported by ONR.]

11:35

**U13. Volume scattering measurements using parametric sonar.** James M. Syck and Charles L. Brown (Code 3112 New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

A parametric sonar has been used to measure volume scattering strength at a number of ocean locations. The narrow beamwidth and absence of sidelobes of the parametric sonar make it an ideal tool for measuring volume scattering strength as a function of depth. The resolution of this device is such that in some circumstances individual scatterers can be identified. An example of such a situation is presented. The parametric sonar system is described. Additional sets of examples from near-shore and off-shore areas are presented for both day and night conditions.

11:50

**U14. A fast and accurate bistatic bottom reverberation model.** D. R. Haller (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada)

A simple bistatic reverberation model has been developed to model low-frequency, wideband, bottom, or surface reverberation. The model allows for arbitrary separation of source and receiver in both range and depth, and accommodates both source and receiver beam patterns. Total reverberation is computed by the incoherent summation of the incremental contributions from each three-dimensional ray path connecting the source, scattering element, and receiver. Simplified ray path calculations are made with either a constant or a linear sound speed profile, assuming

spherical spreading along each ray, and specular reflection from the flat, horizontal boundary interfaces. Acoustic characteristics of the surface and bottom are assumed to be completely represented by the reflection loss and scattering strength functions. A realistic representation of generalized bistatic scattering [D. D. Ellis and D. R. Haller, *J. Acoust. Soc.*

*Am. Suppl.* 1 82, S124 (1987)] has been successfully used to model reverberation from explosive sound sources in the Tufts Abyssal Plain of the Northeast Pacific Ocean. Predictions from the simple reverberation model were found not only to be very similar to those of the bistatic generic sonar model but also to require much less computation time.

WEDNESDAY MORNING, 18 MAY 1988

WEST BALLROOM (A&B), 9:00 TO 11:35 A.M.

## Session V. Architectural Acoustics II and Musical Acoustics II: Concert Hall Acoustics Based on Ando's Work

Richard Talaske, Chairman

*Talaske-Joiner Group, Incorporated, 137 North Oak Park Avenue, Oak Park, Illinois 60301*

Chairman's Introduction—9:00

### *Invited Papers*

9:05

**V1. Further investigation on concert hall acoustics.** Yoichi Ando (Faculty of Engineering, Kobe University, Kobe 657, Japan)

Since the book [Y. Ando, *Concert Hall Acoustics* (Springer, New York, 1985)] was published, further work has been carried out. The results demonstrate that the running autocorrelation function of source signals may better describe preferred conditions of sound fields than does the long-time autocorrelation function. The minimum value of the effective duration of the running autocorrelation function of music signals for the interval used for the subjective preference judgment corresponds well to the preferred initial time delay gap between the direct sound and the first reflection when the amplitude  $A_1 = 0$  dB. This was discovered when Japanese music containing a so-called "MA," a stationary sound passage without any rapid movement, was used as a source signal. Also, it was found that a flat frequency characteristic of the subsequent reverberation time is the optimal condition, and that it may be calculated by the minimum effective duration of the running autocorrelation function of the source signal. Several other results will be presented.

9:35

**V2. Early reactions to Yoichi Ando's book, *Concert Hall Acoustics*.** Leo L. Beranek (BBN Laboratories, 10 Moulton Street, Cambridge, MA 02238)

Ando's book joins one of a handful written since 1900 on the science of concert hall acoustics. It contains seven chapters packed with physical acoustics, the behavior of the physical hearing system, the response of the nervous system, psychoacoustic judgments, the composition of a sound field in a hall, prediction of subjective preferences in halls based on the sound field, and design studies for concert halls. This paper attempts to relate Ando's teachings to the practical experience the author has had with the acoustics of a number of concert halls. Areas in which the book fits these examples well and in which further research seems indicated will be presented.

10:05

**V3. Solved and unsolved problems in concert hall acoustics.** Manfred R. Schroeder (Drittes Physikalisches Institut, University of Goettingen, D-3400 Goettingen, Federal Republic of Germany)

During the last 2 decades, several difficult problems in concert hall acoustics have been solved or elucidated. Among these, the recognition of the importance of early lateral sound ranks high. This piece of progress was made possible by the combined efforts of experts from widely varying disciplines (physicists, psychologists, mathematicians, and—last but not least—the observant consultant). But much remains to be done to strengthen the science of room acoustics further. With digital electronics reaching new pinnacles of sophistication, *active sound field modification* should be pursued much more actively—for multipurpose halls and other spaces. *Diffusion* of radiated sound can be controlled by the same number-theoretic principles that are finding

increasing use in passive designs [M. R. Schroeder, *Number Theory in Science and Communication* (Springer, New York, 1986), 2nd ed.]. But the optimum degree of diffusion requires further *subjective studies*. Generally usable software for *digital modeling* of acoustic spaces should be developed to exploit more fully the capabilities of modern computers (both serial and parallel). Room acoustical measurement methods that use *speech and music as test signals*, to permit measurements during actual performances, should be perfected and used as a matter of course. The future of room acoustics is bright if we grasp the available opportunities.

### Contributed Papers

10:35

**V4. Sound field modeling and simulation of a concert hall.** T. Hidaka, K. Kageyama, and S. Masuda (Takenaka Technical Research Laboratory 5-14, 2-chome, Minamisuna, Koto-ku, Tokyo 136, Japan)

A sound field simulator has been developed in order to subjectively evaluate hall acoustics in the laboratory. This process consists of three parts: (1) a computer program for deriving the impulse response of the room from the architectural drawings using the ray tracing technique; the program is written mainly in FORTRAN and the calculation is implemented using a maximum of 1500 plane surfaces and 1 000 000 rays at most; (2) a new source of orchestral music that is digitally recorded in a temporarily constructed anechoic room for this purpose, and (3) hardware to synthesize the calculated sound field, which convolutes impulse response with input signal by digital signal processing and outputs from 24 speakers set up in an anechoic room in real time. In order to apply this simulator, subjective experience by means of audition for comparison of six concert halls is made and its ability and accuracy are investigated.

10:50

**V5. Effect of transient signal length on interaural cross-correlation functions.** Hirofumi Yanagawa (Department of OMDD, Pioneer Electronic Corporation, 2610 4-chome, Hanazono, Tokorozawa City, Saitama 359, Japan)

The internal cross correlation is closely related to the subjective impressions of sound fields. This paper attempts to account, in a comprehensive way, for the combined effects that initial reflected sound and reverberation sound from music or other transient signals have on such impressions. To this end, cross-correlation functions of the rise and fall of the sound field from transient signals were derived from the windowing impulse responses at two points in a hall. These results were combined with the image-sources distribution patterns derived by the closely located four-point microphone method. Good agreement was found between changes with time in experimentally derived maximal cross-correlation function values and the image-sources distribution of the sound field.

11:05–11:35

### Panel Discussion

PANELISTS: Yoichi Ando  
Leo L. Beranek  
Manfred R. Schroeder

WEDNESDAY MORNING, 18 MAY 1988

GRAND BALLROOM B, 9:30 TO 11:15 A.M.

### Session W. Psychological Acoustics III: Discrimination and Perception of Complex Stimuli

Robert S. Schlauch, Chairman  
*Psychology Department, University of California, Berkeley, California 94720*

### Contributed Papers

9:30

**W1. Directing attention in pitch and time: Effects on the perception of hidden melodies.** W. Jay Dowling (Program in Human Development and Communication Sciences, University of Texas/Dallas, Richardson, TX 75083-0688)

Listeners attended to a target tone in the middle of a familiar melody that was interleaved with distractor notes. The target occurred in either expected or unexpected locations in pitch or in time. Expectancy was

controlled by means of a cue melody preceding the embedded target. In half the sessions, the cue indicated the pitch level of the target, and listeners judged the time at which it occurred. In the other sessions, the cue indicated time, and listeners judged pitch. The cue was 83% valid, and *expected* targets matched the cue while *unexpected* targets did not. Rate of presentation was adjusted within each session to produce approximately 80% accuracy. Temporal expectancies had a much greater effect on reaction times (RTs) than did pitch expectancies. RTs for pitch judgments were about 130 ms longer when targets occurred at unexpected (versus

expected) times, while RTs for time judgments were only about 25 ms longer for targets at unexpected (versus expected) pitches. Rhythmic organization provides a basis for the listener to direct attention to critical events in a rapid auditory stream.

9:45

**W2. The subjective size of musical intervals.** Lyne Plamondon (Department of Psychology, University of California, Berkeley, CA 94720)

The musical scale and the psychophysical scales of pitch appear to be in contradiction with one another. The debate centers on whether or not pitch is a log function of frequency. Previous results on the listeners' judgment of ratios and differences in pitch between pairs of tones agree with the musical scale [R. Elmasian and M. H. Birnbaum, *Percept. Psychophys.* **36**, 531–537 (1984)]. In the present experiment, the subjective size of musical and nonmusical intervals was measured by magnitude estimation and category rating, using subjects without musical training. Pure tones as well as complex tones (harmonic series) were used. Frequencies ranged from 110–2145 Hz. For each pair of tones presented melodically, subjects were instructed to rate the pitch of the second tone relative to the pitch of the first one. Results show that subjects use the same comparison operation for both ratio and difference judgments of pitch. The effects of frequency range, spectral composition, size of interval, and musicality of the ratios on the subjective size of musical intervals are discussed. [Research supported by NIH.]

10:00

**W3. Perception of timing in rhythmic patterns.** Caroline B. Monahan and Ira J. Hirsh (Central Institute for the Deaf, St. Louis, MO 63110)

In two studies employing an adaptive, cued-2AFC procedure, listeners discriminated between six-tone rhythmic patterns that differed only in the delay of the temporal position of one of the tones. On each trial, feedback was given and the subject's performance determined the amount of delay on the next trial. The six tones of the patterns marked off five intervals. In the first study, eight patterns comprised three "short" and two "long" intervals: thus 12121, 21121, etc., where the "long" (2) was twice the length of a "short" (1). In the second study, eight patterns comprised two "shorts" and three "longs": thus 21212, 12212, etc. Each pattern was tested 45 times (five positions of the delayed tone  $\times$  three tempos  $\times$  three replications) for each of three subjects. Consistent with previous work on simple interval discrimination, absolute discrimination ( $\Delta t$  in ms) was poorer, the slower the tempo. Relative discrimination ( $\Delta t/t$  "short," where "short" was 50, 100, or 200 ms) was better, the slower the tempo. Beyond these global results, large interactions of pattern with position of the delayed tone and tempo suggest that different models of performance are needed to explain behavior at the different tempos. [Work supported by AFOSR.]

10:15

**W4. The perception of time-separation pitch by dolphins.** Whitlow W. L. Au (Naval Ocean Systems Center, P. O. Box 997, Kailua, HI 96734) and Jeffrey Pawloski (SEACO Inc., Kailua, HI 96734)

A dolphin was required to discriminate between rippled and nonrippled noise projected by an underwater transducer. Random noise was summed with its delayed replica to produce noise having ripples separated by  $1/T$  Hz in the frequency domain, where  $T$  is the delay time. The dolphin detected the cos - rippled stimulus at a correct response level of at least 75% for delays between 15 and 500  $\mu$ s, and the cos + rippled stimulus for delays of 13 to 190  $\mu$ s. The dolphin's sensitivity to rippled noise was measured by attenuating the delayed replica for different delays. The dolphin was most sensitive for a delay of 100  $\mu$ s. Its sensitivity at 100  $\mu$ s was 5 dB better for the cos + than the cos - stimuli. The broadband cos + noise was also filtered in different 1/3 octave bands to determine if the

animal's sensitivity to rippled noise was a function of the center frequency of the noise. The dolphin's performance was relatively constant as a function of center frequency. The overall results suggest that dolphins may be able to perceive time-separation pitch.

10:30

**W5. Differential frequency sensitivity of humans and monkeys for tone versus vowel stimuli.** Joan M. Sinnott (Department of Psychology, Indiana University, Bloomington, IN 47405)

Humans and monkeys (*Macaca fuscata*) were compared in their abilities to discriminate frequency changes (a) in a 2-kHz pure tone and (b) along a steady-state /i/ to /i/ vowel continuum ( $F1 = 390 - 270$  Hz;  $F2 = 1990 - 2290$  Hz;  $F3 = 2550 - 3010$  Hz). A repeating standard AX procedure was used and stimuli were presented monaurally through the right ear. For the tone stimulus, human and monkey DLs were markedly divergent; at low SLs, the human DL (6 Hz) was about 1/5 that of monkeys (30 Hz) while, at high SLs, the human DL (4 Hz) decreased to about 1/20 that of monkeys (80 Hz), as monkeys became less sensitive. For the vowel stimulus, human and monkey DLs (measured relative to changes in  $F2$ ) were more similar: The human DL (15 Hz) was about 1/3 that of monkeys (45 Hz), and SL functions were parallel and flat for both species. Results indicate that monkeys provide a much better (although not perfect) model of human sensitivity for a complex vowel stimulus than for a simple tone stimulus. [Work supported by NIH.]

10:45

**W6. Speech perception by budgerigars (*Melopsittacus undulatus*): Spoken vowels.** Susan D. Brown, Robert J. Dooling, and Kazuo Okanoya (Department of Psychology, University of Maryland, College Park, MD 20742)

Budgerigars (parakeets) were trained on a same/different task using operant conditioning. Response latencies were used to construct similarity matrices, and multidimensional scaling procedures were then used to produce spatial maps of these speech stimuli. Budgerigars were tested on stimulus sets consisting of natural exemplars of different vowels produced by a number of male and female talkers. In spite of variation among talkers, budgerigars showed evidence of perceptual categories for different vowels as well as for male and female talkers. The two-dimensional perceptual space for budgerigars discriminating among vowels bears a striking resemblance to the familiar  $F1-F2$  vowel space described by Peterson and Barney [*J. Acoust. Soc. Am.* **24**, 175–184 (1952)]. This suggests that budgerigars use first and second formant information in discriminating among spoken, sustained vowels. [Work supported by NIH.]

11:00

**W7. Speech perception by budgerigars (*Melopsittacus undulatus*): Synthetic VOT stimuli.** Robert J. Dooling, Kazuo Okanoya, and Susan D. Brown (Department of Psychology, University of Maryland, College Park, MD 20742)

Budgerigars (parakeets) were trained using operant conditioning procedures to detect changes in a repeating background of synthetic speech sounds. Response latencies for detection were used to construct similarity matrices, and multidimensional scaling procedures were then used to produce spatial maps of these speech stimuli reflecting perceptual organization. Birds were tested on the same alveolar, velar, and bilabial synthetic VOT stimuli previously used to test humans and chinchillas [P. K. Kuhl and J. D. Miller, *J. Acoust. Soc. Am.* **63**, 905–917 (1978)]. Results indicate that budgerigars partition each of these continua into two perceptual categories with an abrupt perceptual change occurring at roughly the same VOT location as observed in humans and chinchillas. It is concluded that the perceptual mechanisms underlying the categorical perception of VOT stimuli are sufficiently general so as to be common to the mammalian and avian auditory systems. [Work supported by NIH.]



## Session X. Speech Communication IV: Neural Networks in Speech Recognition II (Poster Session)

Les Atlas, Chairman

Department of Electrical Engineering, FT-10, University of Washington, Seattle, Washington 98195

## Contributed Papers

All posters will be displayed from 10:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:00 to 11:00 a.m. and contributors of even-numbered papers will be at their posters from 11:00 a.m. to 12:00 noon.

**X1. A study on the speaker-independent feature extraction of Japanese vowels by neural networks.** Toshio Irino and Hideki Kawahara (NTT Basic Research Laboratories, 3-9-11 Midori-cho Musashino, Tokyo 180, Japan)

The feature extraction characteristics of three-layer back propagation neural networks, when applied to speaker-independent vowel/gender recognition tasks, are investigated. Speech samples are monosyllabic vowels extracted from a syllabic data base of 100 Japanese syllables digitized at a sampling rate of 16 kHz spoken by 100 different speakers. Several LPC-based parameters and physiology-based parameters are tested for input representations. The results reveal that three hidden units are necessary to discriminate five Japanese vowels. Close investigation of hidden unit functions reveals that distributed representations of the targets are developed as hidden unit activation patterns. The frequency domain interpretation of the input weights of hidden units, using autocorrelation inputs, shows that the network extracts conventional knowledge on vowel formant structure. The relations between hidden unit activation patterns and descriptive features, as well as their implications to speech perception research, are also discussed.

**X2. Recognition of vowels using artificial neural networks.** Hong C. Leung and Victor W. Zue (Room 36-547, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

This paper is concerned with the application of artificial neural networks to phonetic recognition. This work is motivated by the observation that improved knowledge of feature extraction is often overshadowed by relative ignorance on how to combine them into a robust decision. The goal is to investigate how the self-organizing framework of artificial neural networks can be exploited to enable different acoustic cues to interact. The investigation is couched in experiments that recognize the 16 vowels in American English, using some 10 000 tokens in all phonetic contexts. The tokens were extracted from 1000 sentences spoken by 140 males and 60 females. It was found that, by replacing the mean-squared error metric with a weighted one to train a multilayer perceptron, better recognition accuracy and rank order statistics were consistently obtained. Using the two-layer perceptron in a *context-independent* manner, a top-choice accuracy of 54% was achieved, which compares favorably with results reported in the literature. *Context-dependent* experiments reveal that heterogeneous sources of information can be integrated to improve recognition performance. The top-choice accuracy of 67% is comparable to the average agreement among listeners. Finally, it was found that the rate of improvement on recognition accuracy may be used as a terminating criterion for training, and that reasonable performance can be achieved using as few as 800 training tokens. [Work supported by DARPA.]

**X3. A code in the nodes.** Lynn A. Streeter, Candace A. Kamm (Bellcore, 435 South Street, Morristown), NJ 07960), and Yana Kane-Esrig (Cornell University, Ithaca, NY 14850)

Four two-layer associative networks (one layer of hidden nodes) were trained to classify spectra extracted from naturally produced vowels of one talker, using the back-propagation learning algorithm [Rumelhart *et al.*, *Nature* 323, 533-536 (1986)]. The inputs to the networks were either LPC spectra or "perceptual" spectra, obtained by convolving the Bark-transformed spectrum with an asymmetric filter. The networks trained on LPC input classified test vowels more accurately than the networks trained on the perceptually based spectra. The networks' solutions were examined to determine (a) whether it was possible to characterize the features underlying the solutions and (b) whether the input spectral representation affected the nature of the solutions. Principal components analyses of the activation patterns of the hidden units showed that three dimensions accounted for over 80% of the variance for all networks. For networks trained on "perceptual" spectra, the first three dimensions were highly correlated with  $F2/F1$  ratio,  $F3/F2$  ratio, and  $F3$ , respectively. For LPC input, the first three dimensions correlated with  $F1$ ,  $F3/F2$  ratio, and  $F2/F1$  ratio. Thus these networks classified the vowels using features related to the formant space.

**X4. Complete gradient optimization of a recurrent network applied to /b/, /d/, /g/ discrimination.** Raymond L. Watrous (Siemens RTL, 105 College Road East, Princeton, NJ 08540 and the University of Pennsylvania, Philadelphia, PA 19104), Bruce Ladendorf (Siemens RTL, 105 College Road East, Princeton, NJ 08540), and Gary Kuhn (IDA-CRD, 1 Thanet Road, Princeton, NJ 08540)

A complete gradient optimization technique for connectionist networks with recurrent links is presented. A truncated gradient technique is available in the literature. A network with recurrent links for discrimination of /b/, /d/, and /g/ in the context of following /i/, /a/, or /u/ is designed. The performance of this network as optimized with either of the two techniques is then compared. It will be shown that the complete gradient is exact and simpler, and it will be demonstrated that it leads to superior performance.

**X5. Speech categorization using recurrent networks.** Sven Anderson, John Merrill, and Robert Port (Department of Linguistics, Indiana University, Bloomington, IN 47405)

Several connectionist networks were trained to classify the English syllables ba, da, ga, pa, ta, ka collected from two male and two female speakers. Using a speech preprocessor, perceptually based spectral patterns were computed [H. Hermansky, Proc. ICASSP 87, 1159-1162 (1987)] every 5 ms. A sequential network having a limited class of recurrent connections [M. Jordan, ICS Tech. Rep. University of California at San Diego (1986)] was employed to categorize the data. Training by back propagation or second-order back propagation, a linear increase in the certainty of classification over the course of the syllable was required. Performance of the sequential networks was evaluated on both "known" and "unknown" speakers. When tested on *novel* tokens of a known

speaker, the sequential network did very well as opposed to very poorly on tokens from an unknown speaker. Sequential networks trained with back propagation are capable of integrating cues distributed over time and using them to categorize data. However, their learned behavior may not generalize to data from other speakers. [Work supported by NSF.]

**X6. Effects of network topology on speech categorization.** John Merrill, Sven Anderson, and Robert Port (Department of Linguistics, Indiana University, Bloomington, IN 47405)

Recognizing speech requires the identification and analysis of temporally distributed cues. A system must either examine a sufficiently long window at a single glance or else internally accumulate stimulus information. Sequential networks follow the second path by storing information internally in state nodes. Feed-forward networks do not maintain their history internally and thus require that the speech signal be presented in fixed windows. The performances of sequential and feed-forward networks at recognition of auditorily preprocessed stop-vowel syllables are compared. Several feed-forward networks were trained by presenting a whole syllable to the network as a single token and requiring categorization. The sequential networks were more robust within and across speakers than the feed-forward networks. Unfortunately, the use of the back-propagation algorithm to train a sequential network requires presentation of a desired output *at every time slice*. This forces arbitrary choices for specifying target outputs. The results suggest that techniques based on back-propagation will prove inadequate to train networks to perform speech categorization across variations in speaker. This difficulty could be obviated by employment of a learning paradigm that does not require immediate feedback, such as a stochastic learning algorithm. [Work supported by NSF.]

**X7. Using an adaptive network to recognize demisyllables in continuous speech.** Candace A. Kamm, Thomas K. Landauer, and Sharad Singhal (Bellcore, 435 South Street, Morristown, NJ 07960)

A nonlinear, multilayer associative network was trained on a speech recognition task using continuous speech. Naturally spoken 14-syllable "sentences" from one talker were preprocessed to produce a 15-band spectral representation incorporating several transformations introduced by the peripheral auditory system on acoustic signals. Input nodes to the network represented a 150-ms window through which the spectral representation passed in 2-ms steps. A single layer of 20 hidden nodes was used. Output nodes represented seven initial demisyllables whose target values were specified based on a human listener's identification of the sounds heard during the input segment. The network was trained to criterion using a variant of the back-propagation learning algorithm [Rumelhart *et al.*, *Nature* **323**, 533-536 (1986); Landauer *et al.*, *Proc. Cog. Sci. Soc.*, 531-536 (1987)]. A minimum-error-rate figure of merit (derived from signal detection theory) was used to evaluate the effect of the size of the training corpus on the network's performance. Minimum error rate on a test corpus of demisyllables spoken in different contexts decreased from 5.2% to less than 2% as the number of sentences in the training corpus was increased from one to seven.

**X8. Comparison of standard ASR front ends and auditory models in neural net-based automatic speech recognition.** Mark Terry and Hynek Hermansky (U.S. West Advanced Technologies, 6200 South Quebec Road, Englewood, CO 80111)

Recent work [Renals and Terry, European Conference on Speech Technology, Edinburgh, United Kingdom (1987)] reported on a small multispeaker isolated digit automatic speech recognition (ASR) experiment using a backward propagation neural net system with a synchrony auditory model as its front end. In spite of fairly crude temporal normalization, the system was capable of better than 80% ASR accuracy. The question remains if the reported performance is due to (a) the particular front end, (b) the particular neural net-based classification, (c) the particular time-normalization scheme, or (d) the combination of all factors.

The goal here is to isolate these factors. In the present contribution, different front ends in the neural net-based ASR are systematically evaluated. Several standard ASR front ends are compared with the synchrony auditory model and the perceptually based linear predictive auditory model front ends in both the speaker-dependent and the speaker-independent ASR. The speed of learning and the ASR accuracy of the compared recognizers are reported and discussed.

**X9. Isolated word recognition based on the adaptive neural network.** J. S. Wang and B. Widrow (ISL, Stanford University, Stanford, CA 94305)

For the last several years, despite many new approaches (such as application of the Markov model and VQ) in speech recognition, the classical pattern matching using dynamic programming still prevails because it yields good recognition results. However, this DP-based algorithm requires a large amount of computation. It is difficult to meet the real-time requirement for large vocabularies. An alternative pattern recognition method called the adaptive neuron in adaptive control system used by B. Widrow in the early 1960s is used here to build an isolated word recognition system. A simple and efficient learning algorithm is presented for adaptively adjusting weight vectors to fit a certain word pattern. In order to capture the variations in speech, an increasing codebook was designed during the learning phase to allow increasingly more patterns; therefore, multiple reference patterns were adopted, giving the advantages of a statistical Markov model. The performance of this recognition system has been preliminarily tested on two common vocabularies including ten digits and the English alphabet. The recognition accuracy is 100% for digits after five trainings per word and 93% for the alphabet after ten trainings per word for speaker trained. Further experiments for a complete Chinese vocabulary are under progress.

**X10. An artificial neural network-based codebook search technique.** Charles Pope, Les Atlas, and Charles Nelson (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

An unsupervised learning artificial neural network (ANN) [T. Kohonen, Helsinki University of Technology Report TKK-F-A601 (1986)] was modified for a vector-quantized PCM (VPCM) codebook search problem. The ANN was structured such that each element of the network had associated with it a single codebook vector. The amount of processing required at each element of the network was used to derive an upper bound on the number of network iterations. Simulations were performed to determine the effect that various network parameters had on the speed of network convergence. The parameter values offering the greatest performance were applied to the network. The speed and computational complexity of the ANN solution to this problem were then compared to the same criteria for a standard linear (full-codebook) search technique. Analysis and test results indicate that the ANN approach can provide the speed of a tree search coupled with the minimum memory characteristics associated with a linear search, at the expense of requiring a multiprocessor configuration. [Work supported by NSF.]

**X11. Enhanced speaker identification through the use of cepstral coefficients and redundant time frame elimination.** Timothy R. Thomas, George J. Papcun (Los Alamos National Laboratory, Los Alamos, NM 87545), Stan Willie, and Kalyan Ganesan (U.S. WEST Advanced Technologies, Englewood, CO 80111)

A text-independent speaker identification system was developed on two male and two female speakers with similar accents. Each read a phonetically balanced list of ten sentences. The system was trained repeatedly on a rotated set of nine sentences and tested on the remaining one. Speech during successive 16-ms windowed time slices was described by 14 cepstral coefficients. Unit direction vectors were used to characterize each sentence from each speaker. A nearest-centroid, nearest-neighbor, or improved perceptron neural net training procedure was used to define deci-

sion regions. When the data were preprocessed so as to remove time slices that were similar in all speakers, discriminability was enhanced and errorless identification was obtained. The success of this system appears to

result primarily from the ability of the cepstral coefficients to capture the speaker-dependent information in the higher formants and from the accentuation of this information by the preprocessor.

WEDNESDAY AFTERNOON, 18 MAY 1988

GRAND BALLROOM C, 12:45 TO 2:32 P.M.

## Session Y. Speech Communication V: Speech and Speaker Recognition

Victor W. Zue, Chairman

Department of Electrical Engineering and Computer Science, Room 36-591, Massachusetts Institute of Technology,  
77 Massachusetts Avenue, Cambridge, Massachusetts 02139

Chairman's Introduction—12:45

### Contributed Papers

12:47

**Y1. Comparative study of ASR front-ends in noise.** Jean-Claude Junqua (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105 and C.R.I.N.-C.N.R.S., Vandoeuvre les Nancy, France) and Hisashi Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

In automatic speech recognition (ASR) of speech corrupted by noise, the performance tends to deteriorate rapidly depending on the choice of analysis method and distance measure. In order to evaluate the recognition performance for several analysis methods and distance measures, a series of isolated word recognition experiments was performed. Analysis methods selected are critical-band filtering, perceptually based linear prediction (PLP), linear prediction (LP), and time synchronous linear prediction (SLP). The weighted Euclidean distance with different weightings [unity, root power sums (RPS), and exponential filtering] was applied in the cepstrum domain. Experiments were carried out for clean speech and for two noise conditions (white and low-pass filtered white, added to the clean speech) at different SNR ratios (25 to 5 dB), using an alphanumeric vocabulary (ten speakers). It is shown that improvements in robustness of the recognizer in noise can be achieved by a proper selection of analysis method and cepstral weights used in the front-end. Improvements are found over the RPS distance measure (previously shown to be useful in noise conditions with LP and PLP analyses) [B. Hanson and H. Wakita, *Proceedings ICASSP 86* (IEEE, New York, 1986), pp. 757-760] by use of the general exponential lifter.

1:02

**Y2. Feature-based automatic syllable and stress detection.** Briony Williams and Jonathan Dalby (Centre for Speech Technology Research, University of Edinburgh, 80 South Bridge, Edinburgh EH1 1HN, Scotland)

The importance of syllable structure and stress level as determinants of segmental temporal and spectral variability makes automatic syllable detection and stress estimation a very desirable goal for continuous speech recognition research. In this paper, a rule-based system is described for locating syllables in continuous speech and for making a two-level stress assignment. Location of syllable nuclei and rough estimation of syllable boundaries are performed using a smoothed midfrequency "sonorant" energy contour, a frication detector, and sonorant consonant detectors. First pass classification of detected syllables as stressed, unstressed, or uncertain is based on the relative energy levels and utterance-position-normalized durations of syllables in a three-syllable window. Fundamen-

tal frequency information is then used to reclassify uncertain cases as either stressed or unstressed. Preliminary evaluation of the system's current performance on a multispeaker data base yielded 77% correct location and classification of syllables. Although improvement is necessary, this result is encouraging.

1:17

**Y3. A segmentation algorithm based on spectral variance.** A. Kumar and H. Wakita (Speech Technology Laboratory, 3888 State Street, Santa Barbara, CA 93105)

Presented is a segmentation algorithm based on spectral variance. The speech signal is first segmented into spectrally stable segments by using a median smoothed spectral variance over a 70-ms window. The segment boundaries are placed at the maxima in the spectral variance and the minima give typical frames for the segments. The spectral variance peak for the glides is generally very small because of their smooth transition. Hence, the glides are segmented by using second and third formant trajectories. Extraneous segments outside the word boundaries are eliminated by adaptive silence detector. The segments are then assigned broad phoneme classes by using a tree classifier on the LPC cepstral coefficients for the typical frames. The fricative and nonfricative segments are distinguished by the normalized differenced speech signal. The tree classifier is separately trained on hand-labeled databases. Preliminary experiments show that 90% of the segments are detected. Results for the 104-word keyboard vocabulary for six males and four females and continuous speech will be presented.

1:32

**Y4. Speech recognition using a synthesized codebook.** L. A. Smith, B. L. Scott, R. G. Goodman, L. S. Lin, and J. M. Newell (Scott Instruments Corp., 1111 Willow Springs Drive, Denton, TX 76205)

Speech sounds generated by a simple waveform synthesizer were used to create a vector quantization codebook for use in speech recognition. Recognition was tested over the TI-20 isolated word database using a conventional DTW matching algorithm. Input speech was filtered to limit the bandwidth to 300-3300 Hz, then was passed through the Scott Instruments Coretechs process, implemented on the SI2010 signal processing chip, to create the speech representation for matching. Synthesized sounds were processed in software by an SI2010 emulation program.

SI2010 emulation and recognition were performed on a DEC VAX 11/750. The original codebook contained 109 vectors. This codebook was decimated through the course of the experiments, based on the number of times each vector was used in quantizing the training data for the previous experiment. Recognition scores are presented for progressively smaller codebook sizes, as well as for the baseline condition (no vector quantization).

1:47

**Y5. An efficient, robust speaker-independent algorithm.** Brian Scott, Lisan Lin, Mark Newell, and Lloyd Smith (Scott Instruments Corporation, 1111 Willow Springs Drive, Denton, TX 76205)

The algorithms described have yielded speaker-independent scores of 95.1% on the 20-word TI database obtained from the National Bureau of Standards. Results were obtained by training the system on half of the speakers in the database, testing on the other half, and then reversing the order. Training was done with the 10 training tokens per speaker per word only. Testing was on the 16 test tokens per speaker per word. The total number of test trials was 5120. The recognizer uses conventional methods for time normalization and matching. Time normalization is linear and scoring is accomplished with a simple differencing algorithm weighted by variances. Storage requirement is 3072 bits per word. Most of the speaker normalization is accomplished by the proprietary signal processing method developed by Scott Instruments. Aside from the amplitude normalization routines, no floating point arithmetic is used. All signal processing is temporally based. The front end process can be adapted for use with dynamic time warping algorithms or feature based algorithms. The system is, therefore, extensible to connected speech.

2:02

**Y6. Recognition of continuously spoken letters by listeners and spectrogram readers.** Nancy A. Daly and Victor W. Zue (Room 36-575, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

Because of acoustic similarities between some letters of the alphabet, automatic recognition of continuously spoken letters is a difficult task. The goal of this study is to determine and compare how well listeners and spectrogram readers can recognize continuously spoken letter strings from multiple speakers. The interest in spectrogram reading results is

motivated by the belief that this procedure may help to identify acoustic attributes and decision strategies that are useful for system implementation. Listening and spectrogram reading tests involving eight listeners and six spectrogram readers, respectively, were conducted using a corpus of 1000 wordlike strings designed to minimize the use of lexical knowledge. Results show that listeners' performance was better than readers' (98.4% vs 91.0%). In both experiments, string lengths were determined very accurately (98.1% and 96.2%), presumably due to the large number of glottal stops inserted at letter boundaries to facilitate segmentation. Most of the errors were due to substitution of one letter for another (68% and 92%), and they generally fall into two categories. Asymmetric errors can often be attributed to subjects' disregard for contextual influence, whereas symmetric errors are largely due to acoustic similarities between certain letter pairs. Subsequent acoustic study of four of the most confusable letter pairs has resulted in the identification of a number of distinguishing acoustic attributes. Using these attributes, overall recognition performance better than that of the readers was achieved. [Work supported by NSF and DARPA under contract N00014-82-K-0727, monitored through the Office of Naval Research.]

2:17

**Y7. Human and machine performance on speaker identity verification.** Timothy C. Feustel (Bell Communications Research, 435 South Street, MRE 2E 236, Morristown, NJ 07960), Robert J. Logan (SUNY at Binghamton, Binghamton, NY 13901), and George A. Velius (Bell Communications Research, 435 South Street, MRE 2E 236, Morristown, NJ 07960)

Two experiments were conducted to identify acoustic features for speaker identity verification (SIV) that are used by humans and not by cepstral-based algorithms. Although these algorithms generally outperform human listeners for randomly selected comparisons between single-word utterances, this approach was to analyze human performance on comparisons that could not be effectively discriminated by machine. Experiment 1 showed that humans could perform at high levels of accuracy on these comparisons suggesting that either information exists that is not captured by the algorithms, or that the information is coded by the algorithms but is not used effectively. The second experiment consisted of three stimulus conditions for SIV; digitized speech signals, noise-excited re-synthesized LPC signals, and error prediction signals from the LPC. Results indicated high levels of performance in the natural and error prediction signal conditions and performance near chance in the noise excited condition, thus suggesting that the error signal provides valuable information that allows humans to distinguish between speakers. It may be possible to improve verification algorithms by adapting current models to more accurately utilize information used by human listeners.

**Session Z. Physiological Acoustics II: Auditory-Evoked Responses**

John H. Mills, Chairman

*Medical University of South Carolina, Department of Otolaryngology and Communication Sciences, 171 Ashley Avenue, Charleston, South Carolina 29425**Contributed Papers*

1:00

**Z1. Age-related changes in auditory brain-stem responses of Mongolian gerbil.** John H. Mills, Richard A. Schmiedt, Joe C. Adams, Bradley A. Schulte, and Lawrence R. Kulish (Department of Otolaryngology and Communication Sciences, Medical University of South Carolina, Charleston, SC 29425)

As part of a large study of age-related hearing loss, the Mongolian gerbil as an animal model is evaluated. Part of this effort involves monitoring electrical potentials recorded from the auditory nerve and brain stem. The gerbils are born and raised in an acoustically treated quarters where the median sound level is 35 dBA. Test signals are 1.8-ms tone pips ranging from 1–16 kHz in octave steps. By age 22–24 months, auditory sensitivity is decreased by about 10 dB at most frequencies although there are large differences between animals. By age 36 months, mean threshold changes are largest at 8 and 16 kHz, about 35 dB, and decrease to 10–20 dB at lower frequencies. The mean data for the 36-month group form an audiometric configuration that is qualitatively similar to that observed for 60- to 65-year-old human males and 70-year-old human females. Individual differences are large with some animals having nearly normal hearing and others having hearing losses of 50–70 dB at all test frequencies. [Work supported by NINCDS PO NS-25039.]

1:15

**Z2. Auditory responses to the envelopes of pseudorandom noise stimuli in humans.** Robert A. Dobie and Michael J. Wilson (Department of Otolaryngology, RL-30, University of Washington, Seattle, WA 98195)

Averaged scalp potentials evoked by continuous pseudorandom noise can be cross correlated with the evoking stimulus, yielding a cross-correlation function (CCF) that reflects neural phase locking and is quite sensitive for low-frequency stimulus components [M. J. Wilson and R. A. Dobie, *Electroencephalogr. Clin. Neurophysiol.* **66**, 529–538 (1987)]. However, for higher frequency signals, replicable CCFs can only be obtained at moderate to high intensity. Since auditory neurons also respond to envelopes of complex sounds, even for high-frequency carriers, scalp responses were compared to the envelopes of complex sounds; the resultant envelope cross-correlation functions (ECCFs) contained replicable response components primarily below 1000 Hz, regardless of the evoking stimulus spectrum. The ECFF thresholds for three octave-band stimuli (830–1562, 1611–3125, and 3174–6201 Hz) were more sensitive than CCF thresholds ( $p = 0.018$ ), averaging 35-dB spectrum level for ten normal subjects. When stimuli with only odd harmonics were used, replicable odd-component responses were seen only in the spectral range of the stimuli, while even-component responses (presumably to stimulus envelope) were seen only in the low-passed scalp responses.

1:30

**Z3. Human auditory brain-stem asymmetry in the frequency following response (FFR).** Ballachanda B. Bopanna and George Moushegian (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

The existence of cortical asymmetries in humans is well documented and paralleling this is the suggestion that speech sounds are processed by different central neural pathways than nonspeech sounds. This study presents data that reveal lower brain-stem asymmetry to a low-frequency sinusoidal. Frequency following responses (FFR) (500 Hz) were obtained from ten normal hearing subjects monaurally and binaurally over a range of intensities. The results provide clear evidence that the monaural FFRs have differences in onset latency and differences in latencies throughout the response waveform as well as diversities in their amplitudes. Since the FFR emanates from sites that are not rostral to the inferior colliculus, the findings support the view that the lower auditory brain stem is functionally asymmetrical.

1:45

**Z4. The effect of stimulus repetition rate on interaural ABR latency differences in normal and hearing-impaired subjects.** Maureen Hannley, Lorene Weichert, and Sarah B. Wedekin (Division of Otolaryngology and Head/Neck Surgery, Stanford University Medical Center, Stanford, CA 94305)

Twenty-five subjects with normal hearing and 15 with bilateral high-frequency sensorineural hearing impairment were presented click stimuli at four repetition rates (11, 33, 67, and 89 clicks/s) to assess interaural symmetry of wave V latency and amplitude. Predictably, amplitude was significantly decreased and latency increased across all subjects at the higher repetition rates. In both groups, however, interaural latency differences (ILDs) remained with the 0.3-ms normal criterion previously established for slow repetition rates. Thus, both abnormal repetition rate effects and interaural latency differences can serve as reliable indicants of retrocochlear auditory dysfunction, even when peripheral hearing impairment exists.

2:00

**Z5. The auditory brain-stem response in children with a history of OME.** Ron D. Chambers, Lynne E. Rowan (Department of Speech and Hearing Science, University of Illinois, 901 South Sixth Street, Champaign, IL 61820), and Michael A. Novak (Carle Clinic Association-4W, 602 West University, Urbana, IL 61801)

The auditory brain-stem response (ABR) was used to evaluate the auditory-neural status of children with a history of frequent or persistent otitis media with effusion (OME). Subjects were selected from the caseload of a board certified otolaryngologist and from the population at large. They were pair-matched by age and sex to form an otitis-prone and a control group. The otitis-prone group included only those children whose degree of OME involvement was sufficient to warrant tympanostomy tube placement. Stimuli were rarefaction clicks presented at 85 dB nHL at a rate of 21.4/s. The latency of wave III and the III–I interwave interval were longer for the otitis-prone subjects than for the control subjects, which agrees with earlier results [R. C. Folsom, B. A. Weber, and G. Thompson, *Ann. Otol. Rhinol-Laryngol.* **92**, 249–253 (1983)]. The results and precautions in interpretation will be presented. [Work supported by Carle Research Foundation.]

2:15

**Z6. Intraoperative variability of auditory brain-stem responses during cranial nerve microvascular decompression.** Maureen Hannley (Division of Otolaryngology, Stanford University Medical Center, Stanford, CA 94305), Jer-Min Huang (Department of Physiology, Louisiana State University Medical School, New Orleans, LA 70112), and Peter A. Raudzens (Barrow Neurological Institute, Phoenix, AZ 85001)

The auditory brain-stem response (ABR) recordings of 40 patients who underwent microvascular decompression of cranial nerves V and VII

were analyzed. The latencies of waves I, V, and the I-V interpeak interval were examined as a function of subject age, subject gender, operative site, and operative stage. The results indicated that: (1) subject age was not a significant factor in intraoperative ABR variability; (2) significant male-female latency differences were present only in the earlier operative stages; (3) greater latency changes were produced during decompression of the facial nerve than during trigeminal nerve decompression; and (4) greater response variability was present during the later operative stages. The patients' postoperative auditory status was studied to determine the predictive efficiency of three currently used criteria for significant intraoperative latency change.

WEDNESDAY AFTERNOON, 18 MAY 1988

WEST BALLROOM A, 1:20 TO 2:50 P.M.

### Session AA. Physical Acoustics IV: Miscellaneous Topics in Physical Acoustics

Steven R. Baker, Chairman

*Physics Department, Code 61BA, Naval Postgraduate School, Monterey, California 93943*

#### *Contributed Papers*

1:20

**AA1. Evolution of unstable disturbances at a mean-flow stagnation point.** Charles Thompson and Martin Manley (Department of Electrical Engineering, University of Lowell, 1 University Avenue, Lowell, MA 01854)

The interaction between infrasonic, acoustic disturbance and the mean flow near the stagnation point of a bluff body will be examined. The criterion for stability and intermodulation of the disturbances will be discussed. The spatial and temporal evolution will also be addressed. It will be shown that infrasonic and unstable vortical disturbances, working in conjunction, can serve to modulate the amplitude of the acoustic wave. For a harmonically time-varying acoustic wave, this modulation is manifested in an increase in the spectral bandwidth of the wave. [Work supported by Analog Devices Professorship.]

1:35

**AA2. Effect of internal relaxation on the optoacoustic signal in CS<sub>2</sub>.** Charles H. Thompson, Henry E. Bass, and Richard Raspet (Physical Acoustics Research Laboratory, University of Mississippi, University, MS 38677)

The optoacoustic signal in CS<sub>2</sub> has been observed using an experimental configuration similar to that of Sullivan and Tam [B. Sullivan and A. C. Tam, *J. Acoust. Soc. Am.* **75**, 437-441 (1984)] and has been compared to theory and similar results in propanol. The comparisons illustrate that an internal relaxation process is affecting the observed waveform. Assuming a simple three-level system for internal energy and using independently measured vibration-to-translation transfer rates leads to the conclusion that the relaxation process observed is associated with electronic-to-translation or electronic-to-vibration energy transfer with a relaxation time of 90 ns. [Work supported by the Office of Naval Research.]

1:50

**AA3. Photoacoustic method for measuring high-order multiphoton absorption in transparent materials.** Scott C. Jones (Department of Physics, Washington State University, Pullman, WA 99164)

A photoacoustic technique for measuring multiphoton absorption from sharply focused pulsed laser beams is developed for arbitrary order of nonlinear absorption. The resulting sound source is neither spherical nor cylindrical, and a multipole expansion is presented through quadrupole terms for thermal and electrostrictive source terms. The analysis shows that the acoustic amplitude is proportional to absorbed energy, and proportional to the *m*th power of incident laser energy under conditions of *m*-photon absorption. The calibrated measurement of the four-photon absorption cross section of NaCl at wavelength 532 nm is presented. [Work supported by NSF.]

2:05

**AA4. Acoustic imaging by a method of image projections.** Daniel Rouseff, Robert P. Porter, and Leung Tsang (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98125)

A method of three-dimensional acoustic imaging based on *image projections* is derived. A weakly scattering object is illuminated by a plane wave and the scattered field is recorded. Inversion methods based on projections that operate directly on the scattered field are inaccurate due to diffraction effects. By backpropagating the scattered field onto a plane in the image region, diffraction effects are reduced and a projection of the object is generated. Conventional x-ray inversion techniques may then be used to image the object. The relationship between the backpropagated field and the projection is derived both for high-frequency incident waves and low spatial frequency scatterers. Backpropagation additionally allows the use of curved or misaligned recording surfaces. [Work supported by ONR.]

**AA5. Propagation of sound in vibrationally excited  $N_2/He$ ,  $N_2/H_2$ ,  $N_2/CH_4$ , and  $N_2/H_2O$  mixtures.** F. Douglas Shields and L. Dwyann Lafleur (National Center for Physical Acoustics, P.O. Box 847, University, MS 38677)

Previous measurements of resonant reverberation of sound in  $N_2/H_2$  mixtures in a closed tube have shown an amplification of the sound following the rapid excitation of the gas by an electric discharge [J. Acoust. Soc. Am. **81**, 87 (1987)]. These measurements have been extended in this paper to include  $N_2/He$ ,  $N_2/CH_4$ , and  $N_2/H_2O$  mixtures. Some of the excessively large amplification previously observed in  $N_2/H_2$  mixtures has now been attributed to nonlinearity in the microphone response. However, even after correcting this error the amplification is much larger and lasts for a longer time than theoretically predicted. The variation in the translational temperature following the discharge is monitored by measuring the sound velocity. Relaxation times determined from the temperature versus time curves differ only slightly in magnitude and/or tem-

perature dependence from previously reported values. [Work supported by ONR.]

**AA6. Instability and particle agglomeration in acoustically driven flows.** Charles Thompson and Vineet Mehta (Department of Electrical Engineering, Laboratory for Advanced Computation, University of Lowell, 1 University Avenue, Lowell, MA 01854)

The mechanism responsible for instability and turbulence generated by high amplitude acoustic waves has been of interest of in acoustic agglomeration of particles. When a low-frequency oscillation is used in conjunction with a higher frequency acoustic wave, localized unstable boundary layer disturbances can be used to drive the outer flow field. By virtue of the Reynolds stress, the harmonic spectrum of the unstable vortical disturbance serves to modulate the acoustic wave amplitude. This process enhances the ability of the acoustic wave to agglomerate particles. A theoretical model of the instability mechanism and the interaction process will be presented. [Work supported by Analog Devices Professorship.]

WEDNESDAY AFTERNOON, 18 MAY 1988

ASPEN ROOM 1:30 TO 2:30 P.M.

### Session BB. Psychological Acoustics IV and Education in Acoustics III: Auditory Demonstrations on Compact Disc

W. Dixon Ward, Chairman

*Hearing Research Laboratory, University of Minnesota, 2630 University Avenue, S.E., Minneapolis, Minnesota 55414*

#### *Invited Paper*

1:30

**BB1. Auditory demonstrations on compact disc.** A. J. M. Houtsma (Institute for Perception Research, P.O. Box 513, 5600 MB, Eindhoven, The Netherlands), T. D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115), and W. M. Wagenaars (Institute for Perception Research, P.O. Box 513, 5600 MB, Eindhoven, The Netherlands)

In 1984, the Committee on Education in Acoustics of the ASA discussed the possibility of reissuing the "Harvard Tapes," a series of audio demonstrations on tape cassettes originally published in 1978 by David M. Green at the Harvard Laboratory of Psychophysics. The present authors undertook the job of evaluating and updating the content of the original tapes, producing new audio material, and writing a new explanatory book. The compact disc (CD) medium was chosen to replace magnetic tape. Through substantial cooperation from Eindhoven University of Technology, Northern Illinois University, the Philips Company, and the ASA, a CD with a 92-page booklet was developed containing groups of auditory demonstrations divided over seven general topics. For each group, a general introduction with basic references is provided in the booklet. For each demonstration within a group a detailed background, explanation and further literature references are given. Verbal introductions on the CD have been kept very brief and serve mainly to identify each demo. Demonstrations or parts thereof can be called individually in any order, or preprogrammed in arbitrary groups, through 80 separate track or "title" numbers. One thousand copies of the CD set have been made on a first production run. They are available through the headquarters of the ASA.

A CD player with the auditory demonstration will be available at the registration desk for the entire meeting.

## Session CC. Underwater Acoustics IV: Object Scattering

Stephen A. Reynolds, Chairman

*Applied Physics Laboratory, University of Washington, 1013 N. E. 40th Street, Seattle, Washington 98105*

Chairman's Introduction—1:30

## Contributed Papers

1:35

**CC1. Pulse scattering from an object submerged in the ocean.** Michael D. Collins and Michael F. Werby (NORDA Numerical Modeling Division, NSTL, MS 39529)

The low grazing angle asymptotic limit, which is used to derive the parabolic equation, is used to derive an efficient computational method for propagation and scattering of acoustic pulses in the ocean. In order to avoid the time-consuming frequency decomposition, all calculations involving the acoustic field are done in the time domain. The time domain equivalent of the parabolic equation [B. E. McDonald and W. A. Kuperman, *J. Acoust. Soc. Am.* **81**, 1406-1417 (1987)] is used for propagation of the incident field up to the scatterer and for propagation of the scattered field away from the scatterer. Waterman's extended boundary condition, which is a frequency domain method, is used to construct a time domain integral operator that transforms incident waveforms into scattered waveforms and depends only on the geometry and composition of the scatterer. Like the parabolic equation method, this method offers an attractive combination of accuracy and efficiency. [Work supported by ONR and NORDA.]

1:50

**CC2. Principal curvatures of general wavefronts and of reflecting or refracting surfaces.** Cleon E. Dean, W. Patrick Arnott, and P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164-2814)

A nonparametric derivation of the exact form of the two principal curvatures  $\kappa_{1,2}$  of general wavefronts and reflecting or refracting surfaces was made for Cartesian and polar coordinates. The standard expression for the Gaussian curvature  $\kappa_1\kappa_2$  was recovered in the Cartesian case. For the polar case let

$$H = \frac{1}{r} W_r W_r + \frac{1}{r^2} (W_{rr} W_{\theta\theta} - W_{r\theta}^2) + \frac{2}{r^3} W_{r\theta} W_{\theta} - \frac{1}{r^4} W_{\theta}^2,$$

$$B = \frac{1}{r} W_r^3 + \frac{1}{r^2} W_r^2 W_{\theta\theta} + \frac{2}{r^3} W_r W_{\theta}^2 - \frac{2}{r^2} W_r W_{\theta} W_{r\theta}$$

$$+ \frac{1}{r^2} W_{rr} W_{\theta}^2 + W_{rr} + \frac{1}{r} W_r + \frac{1}{r^2} W_{\theta\theta},$$

and  $C = 1 + W_r^2 + (1/r^2) W_{\theta}^2$ , where  $z = W(r, \theta)$  is the polar equation of the surface and  $W_r = \partial W / \partial r$ ,  $W_{rr} = \partial^2 W / \partial r^2$ , and so forth; the analysis gives  $\kappa_{1,2} = [B \pm (B^2 - 4HC)^{1/2}] / (2C^{3/2})$ . The Gaussian curvature is  $\kappa_1\kappa_2 = H/C^2$ ;  $\kappa_1$  and  $\kappa_2$  reduce to previous results [D. G. Burkhard and D. L. Shealy, *Appl. Opt.* **20**, 897-909 (1981)] in the special case  $z = W(r)$ . These results should be useful for generalized ray tracing and for locating caustics in catastrophe acoustics and scattering calculations. [Work supported by ONR.]

2:05

**CC3. Harmonic angular perturbation of a toroidal wavefront: A simple unfolding of an axial caustic.** W. Patrick Arnott and Philip L. Marston (Department of Physics, Washington State University, Pullman, WA 99164-2814)

Axial caustics can be observed in backward and forward scattering from highly symmetric surfaces such as spheres. A wavefront that is locally toroidal, propagates to produce an axial caustic. The unfolding of the axial caustic generated by a harmonic perturbation of the wavefront shape in the azimuthal direction will be considered. The perturbed wavefront shape  $W$  is given by  $W(s, \psi) = [(s - b)^2 / 2\alpha] + \Lambda \cos(2\psi)$ , in polar coordinates  $(s, \psi)$ , where  $\alpha$  and  $b$  specify the characteristic radii of the unperturbed torus and  $\Lambda$  is a measure of the perturbation. The caustic associated with  $W$  consists of four connected transverse cusps in a distorted diamond shape known as an asteroid. The caustic shape has been identified for observation points from the near zone to the far zone. Locations on  $W$  that contribute rays near a cusp point, have been identified analytically. Observations of the asteroid caustic have been made by scattering light from freely rising oblate gas bubbles in water. The bubbles are oblate since they are larger than those considered in our previous study of backscattering [W. P. Arnott and P. L. Marston, *J. Opt. Soc. Am. A* (in press)]. The far zone scattering exhibits an asteroid caustic decorated by a diffraction pattern. The calculated caustic agrees favorably with observations. This calculation should also be useful in acoustic reflection and scattering problems. [Work supported by ONR.]

2:20

**CC4. Comparison of asymptotic and exact results for scattering from rigid spheroids.** Michael F. Werby and Michael D. Collins (NORDA Numerical Modeling Division, NSTL, MS 39529)

Results generated with the Kirchhoff approximation, which is effective for backscattering, and the diffraction integral method, which is effective for forward scattering, will be compared with the exact solution obtained with Waterman's extended boundary condition method. Calculations will be presented for  $kL/2$  ranging from 5 to 300 and for aspect ratios ranging from 1 to 15. An additional asymptotic approximation, the method of stationary phase, is used for higher frequencies. For higher aspect ratio targets, one would expect the accuracy of high-frequency methods to be difficult to predict and to depend not only on  $kL/2$  but also on bistatic configurations due to the large variation of Gaussian curvature on these targets. Hence, the focus of the presentation will be to illustrate how accuracy depends upon  $kL/2$ , angle of incidence, and angle of observation for high aspect ratio targets. [Work supported by ONR and NORDA.]



**CC5. The effect of an object in a waveguide on the field produced by a point source.** Guy V. Norton and M. F. Werby (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

When the acoustic wave produced by a point source in a shallow water waveguide ensonifies an object, the object acts as a secondary source and in turn generates a guided wave. The effect that the object has on the initial incident field is estimated. In particular, comparisons are made with transmission loss versus range for the point source by itself and with transmission loss versus range for the object as ensonified by the point source. To obtain the scattered field from the object, the scattered field is first generated in the vicinity of the object using a transition matrix that relates

the incident field to the scattered field in a waveguide. The transition matrix is obtained from the extended boundary condition (EBC) method. This solution is coupled with the waveguide solution [G. V. Norton and M. F. Werby, "Some Numerical Approaches to Describe Acoustical Scattering from Objects in a Waveguide," in Proc. Sixth Int. Conf. Math. Modeling, St. Louis, MO, 4-7 Aug. 1987]. This method satisfies all appropriate boundary conditions and yields a continuous solution throughout the waveguide. The object used is a rigid spheroid of aspect ratio 5:1. The frequencies are 100 and 450 Hz. The source, receiver, and object depths are different combinations of 50, 75, and 200 m, and two waveguide water depths at 150 and 400 m are considered. The nature of the effect on the initial incident field due to the object and as influenced by the relative placement of the source and receiver relative to it is demonstrated. [Work supported by NORDA.]

WEDNESDAY AFTERNOON, 18 MAY 1988

EAST BALLROOM B, 1:45 TO 3:00 P.M.

### Session DD. Architectural Acoustics III: Topics in General Architectural Acoustics

Roy L. Richards, Chairman

*Towne, Richards & Chaudiere, Inc., 105 N.E. 56th Street, Seattle, Washington 98105*

#### *Contributed Papers*

1:45

**DD1. Measuring sound transmission through suspended ceilings.** J. D. Quirt and R. E. Halliwell (Acoustics Section, Institute for Research in Construction, National Research Council of Canada, Ottawa K1A 0R6, Canada)

In most modern office buildings, interoffice partitions stop at the suspended ceiling. Sound insulation between such offices is generally inadequate for speech privacy. This paper presents the first stages of a project to develop practical acoustical solutions. A new laboratory has been constructed; this simulates a pair of adjacent offices with a common plenum space above the suspended ceiling. Characteristics of the test environment and pertinent standard test procedures have been studied. A series of suspended ceiling systems has been tested. The results indicate the relative importance of major physical factors controlling sound transmission through typical suspended ceilings.

2:00

**DD2. Fuzzwall: Blocking sound transmission above suspended ceilings.** J. D. Quirt and R. E. Halliwell (Acoustics Section, Institute for Research in Construction, National Research Council of Canada, Ottawa K1A 0R6, Canada)

Interoffice partitions commonly stop at the suspended ceiling. The plenum space above the ceiling is used as an air-return for the ventilation system. Sound transmission through this space often limits interoffice speech privacy. This paper reports the performance of a barrier system intended to reduce sound transmission without preventing airflow. Fibrous absorptive material has been used as a barrier system. Dependence of the noise reduction on physical parameters (flow resistance, barrier

thickness, density, etc.) and the effect of providing for airflow have been examined. Laboratory results and preliminary field test data will be presented.

2:15

**DD3. Test results for floor and ceiling modifications for impact isolation.** Edward A. Daly (Daly Engineering Company, 11855 S. W. Ridgecrest Drive, Beaverton, OR 97005) and Kerrie G. Standlee (Van Gulik/Oliver, Inc., 543 Third Street, Lake Oswego, OR 97034)

The impact solution evaluation methods presented by the two authors in a paper given at the 112th Meeting of ASA [J. Acoust. Soc. Am. Suppl. 1 80, S56 (1986)] were used to evaluate modifications of floor/ceiling systems. Tests were conducted on the presently most-used floor structure, the wood/concrete structure. Some of the floor modifications produced significant impact sound isolation improvements. However, the floor modifications that produced the best sound attenuations provided unstable flooring conditions. All ceiling modifications produced significant impact sound isolation improvements. The type of construction required for all ceiling modifications were found to be acceptable to the construction trades. An evaluation method developed by one of the authors was used to produce a single number effectiveness rating for the floor/ceiling modifications. The results of these tests indicate ceiling modifications are more effective than floor modifications in providing impact isolation.

2:30

**DD4. The measurement of acoustic reflection coefficients by using cepstral techniques.** J. Stuart Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, West Lafayette, IN 47907)

The author has recently developed a new free field technique for the *in situ* measurement of acoustic reflection coefficients by the use of transient acoustic signals [J. S. Bolton and E. Gold, *J. Sound Vib.* **110**(2), 179–202 (1986)]. The measurement is conducted by using an omni-directional loudspeaker to generate a test signal; a microphone placed between the loudspeaker and the reflecting surface is used to detect both the incident (i.e., the direct) and reflected signals. Cepstral processing was used to eliminate some of the difficulties associated with existing reflection measurement methods. Specifically, in the new technique the direct and reflected signals are measured simultaneously and are allowed to overlap in time, thus relaxing geometrical constraints. Further, if the incident signal is sufficiently broadband, a knowledge of its spectrum is not required nor is it necessary to measure it individually. The technique is based on the direct extraction of the impulse response of the reflecting surface from the power cepstrum of a signal containing both the direct and reflected signals. Full-scale acoustical trials of this technique will be described and measurements of the reflection coefficients of samples of polyurethane foam will be shown.

2:45

**DD5. A boundary integral equation method for transient problems in architectural acoustics.** D. K. Holger (Department of Engineering Science and Mechanics, 2019 Black Engineering Building, Iowa State University, Ames, IA 50011)

A numerical method for solving transient interior acoustic field problems by combining boundary integral equation and inverse Fourier transform methods is described. An indirect boundary integral equation formulation of the Helmholtz equation using hybrid layer potentials is combined with a numerical inverse Fourier transform to obtain time domain solutions for acoustic field variables. Frequency domain representation of transient sources, frequency domain windowing of the integral equation solutions, and frequency domain resolution enhancement are discussed. Numerical results are presented for the acoustic field inside a rectangular prism with various boundary conditions and for various transient source excitations.

WEDNESDAY AFTERNOON, 18 MAY 1988

GRAND BALLROOM C, 3:00 TO 5:00 P.M.

### Plenary Session

Chester M. McKinney, Chairman

*President, Acoustical Society of America*

### Business Meeting

Proposed Amendment of the Bylaws of the Society

### Presentation of Awards

R. Bruce Lindsay Award to Gilles A. Daigle  
Gold Medal to Arthur H. Benade  
Gold Medal to Richard K. Cook

### Invited Speaker

The Honorable Guy Strutt will talk about his grandfather, Lord Rayleigh.

### Musical Demonstration

Stu Dempster of the University of Washington Music Department will demonstrate a didjeridu, an aboriginal Australian musical instrument.

**Session EE. Engineering Acoustics IV: Laboratory and Measurement Condenser Microphones**

George S. K. Wong, Chairman

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

Chairman's Introduction—8:00

*Invited Papers*

8:05

**EE1. Electret condenser microphones in physical acoustics research.** S. L. Garrett (Physics Department, Naval Postgraduate School, Monterey, CA 93943)

The simplest condenser microphone in terms of construction and use is a charged, aluminized Teflon membrane placed directly over a conducting back plate. In this configuration, the low-frequency open circuit sensitivity is determined approximately by the product of the "effective" electret bias voltage and the compressibility of the fluid trapped between the membrane and the backplate, typically 1 mV/Pa in air. This talk will describe the procedures for calibration of such a transducer in a resonator and its application to the measurement of constituents in binary or pseudobinary gas mixtures. Other applications to low-temperature physics will be described that include acoustic measurement of temperature fluctuations with a perforated electret membrane having a detection threshold of  $\approx 10$  nanokelvin/ $\sqrt{\text{Hz}}$  in superfluid  $^4\text{He}$  at 1 deg above absolute zero, and the measurement of the temperature dependence of the order parameter of superfluid  $^3\text{He-A}$  and  $^3\text{He-B}$  at temperatures of  $\frac{1}{1000}$  of a degree above absolute zero. [Work supported by the Office of Naval Research.]

8:25

**EE2. Laboratory calibration of standard and measurement condenser microphones.** Victor Nedzelnitsky (National Bureau of Standards, Building 233 (Sound), Room A149, Gaithersburg, MD 20899)

The reciprocity technique is the most highly developed and most widely used primary calibration method at the major national standards laboratories of the world. Critical measurements of sound pressure in air are usually traceable to such calibrations, which are briefly described with reference to specific apparatus and procedures at the NBS. Secondary methods, including those based upon reciprocity, can offer improved convenience with minor to significant sacrifices in accuracy and precision. Dynamic signal analyzers incorporating digital signal processing are now sufficiently accurate and precise to offer promise for improved and/or more readily automated measurements, especially secondary free-field calibrations. Specific examples from the NBS research and measurement services are discussed.

8:45

**EE3. Microphone comparison with the three-port two-microphone cavity.** George S. K. Wong and Tony F. W. Embleton (Division of Physics, National Research Council Canada, Ottawa, Ontario K1A 0R6, Canada)

The performance of the three-port two-microphone cavity [Wong and Embleton, *J. Acoust. Soc. Am.* **71**, 1276–1277 (1982)] for the comparison of condenser microphones is described. The experiment involves 1-,  $\frac{1}{2}$ -, and  $\frac{1}{4}$ -in. microphones from three manufacturers. With a reference microphone, the above cavity is able to measure, by comparison, the sensitivities and frequency response of various sizes of microphones under different physical conditions, such as with or without protective grids. For frequency response measurements over the frequency range from 20 Hz to several kilohertz, the experimental results are similar to those obtained with the electro-actuator method, and the cavity measurement does not require any access to the air space that is in close proximity to the microphone diaphragm. In general, the frequency response performance of the cavity improves with the smaller size microphones.

9:05

**EE4. Scattering of sound by finite cylinders—A preliminary study applicable to microphones.** A. F. Seybert, M. L. Lin, and T. W. Wu (Department of Mechanical Engineering, University of Kentucky, Lexington, KY 40506-0046)

When the wavelength of sound is of the same order as the diameter of a microphone, the diffraction of sound in the nearfield of the microphone will alter the indicated sound-pressure amplitude. This fact has long been

observed experimentally, and the effect of diffraction is considered in the design of microphones. However, when a pair of microphones is used, as with sound intensity measurements, diffraction alters the amplitude and phase difference indicated by the microphones, and these effects become important at frequencies within the normal operating frequency range of a single microphone of the same diameter. It has been difficult to predict diffraction effects from analytical models, due to the complex geometry involved. A numerical method has been developed, based on the boundary element method, to study the scattering of sound by obstacles of arbitrary shape. As a preliminary model of a microphone, the finite cylinder was used. In this paper numerical results are presented for scattering from a single finite cylinder and from a pair of cylinders. [Work supported in part by the NSF.]

9:25-9:40

Break

9:40

**EE5. Intensity measurement microphones and their calibration.** Erling Frederiksen (Bruel & Kjaer Development, Naerum Hovedgade 18, DK-2850, Naerum, Denmark)

New microphones especially designed for intensity measurements have made it possible to make valid intensity measurements under sound field conditions that previously would lead to invalid results. The achieved improvements are most important at low frequencies but may also be significant, for instance, at 500 Hz, depending on the sound field. The microphones of this new type are partly characterized by a stability and spread in their low-frequency phase characteristics that are a factor of 10-15 times better and partly by a sensitivity to sound pressure at the static pressure equalization vent that is several orders of magnitude lower than that of conventional microphones. These properties are all important for the improvement of intensity instruments that are based on microphones operating on the pressure principle, as the phase difference between exactly spaced points in sound fields need to be measured with great accuracy. System phase errors of the order 0.01 deg are of interest and seem to become possible by proper calibration and correction. The use of conventional microphones implies significant errors due to pressure gradients errors that are eliminated by the extremely low vent sensitivity of the new microphones. Most leading laboratories have spent great efforts on magnitude calibration of microphone sensitivity but very little on phase calibration. The microphones and the methods used for verification of microphone and probe properties are described in this paper.

### Contributed Papers

10:00

**EE6. Study on the linearity of laboratory standard condenser microphone.** S. Yoshikawa (Kanagawa Institute of Technology, Kanagawa, 243-02 Japan)

Because of the many features such as wide frequency range, high stability, good transient response, etc., the condenser microphone has been used as a laboratory standard microphone and a measurement microphone. One problem is that the condenser microphone has essentially nonlinear characteristics on the electrical output/acoustical input function. As sound-pressure level becomes higher, nonlinearity becomes larger. In this study, linearity of the laboratory standard condenser microphone at high sound-pressure level was investigated. In the case of the 1-in. condenser microphone MR-112, linearity was ensured up to 120 dB SPL. Over 120 dB SPL, output/input function becomes larger and distortion becomes detectable. By experiment, measured distortion factor was 2% at 151 dB SPL and 5% at 154 dB SPL. The nonlinearity at high sound-pressure level was studied analytically, and the effect of some parameters such as shape of back electrode, biasing voltage, and stray capacitance was clarified. Based on the analytical study, the possibility of lowering the nonlinearity was investigated. A method to realize the condenser microphone having low distortion and high sensitivity was suggested.

10:15

**EE7. Measurement accuracy of a one-dimensional four-microphone sound intensity probe.** Hideo Suzuki, Masazo Anzai, and Takahiko Ono (Ono Sokki Co., Ltd., 2-4-1 Nishishinjuku Shinjuku-ku, Tokyo 163, Japan)

In the sound intensity measurement, it is important that a microphone probe does not disturb the sound field. A side-by-side or a face-to-face

configuration is most commonly used for the microphone probe. With these conventional configurations, however, two tubes containing individual preamplifiers are parallel and close to each other and cause a diffraction problem. A recently developed one-dimensional four-microphone probe is structurally much simpler than conventional probes. Two microphone pairs cover low- (50-800 Hz) and high- (800-10 kHz) frequency ranges and, therefore, it is possible to measure the whole frequency range without changing the microphone distance. The effective distance between pair microphones, pressure, particle velocity, and active intensity responses are measured in an anechoic chamber. Assuming a farfield condition, the measurement accuracies of pressure, particle velocity, and intensity responses are checked by comparing these with the pressure of a reference microphone. The accuracy of the reactive intensity measurement is checked by measuring the pressure square gradient by a reference microphone. The results will be shown at the meeting.

10:30

**EE8. Mounting and selection of microphones for two-microphone impedance measurements.** Gordon Ebbitt and Jørgen Christensen (Bruel & Kjaer, Naerum Hovedgade 18, DK-2850 Naerum, Denmark)

The determination of acoustic impedance as specified in ASTM E-1050 requires the measurement of sound pressure at two positions along a standing wave tube. The choice of microphones and their mounting can have a significant influence on the measurement results. A variety of microphone types (i.e., phase-matched condensers, probe tube types, condenser microphones fitted with phase correctors, etc.) will be considered and the results from these various types will be compared. The mounting of the microphones (i.e., flush with the tube walls, near the middle of the tube, etc.) will also be considered.

10:45

**EE9. Field calibration and verification of sound intensity systems.** Gordon Ebbitt and Erling Frederiksen (Brüel & Kjær, Naerum Hovedgade 18, DK-2850 Naerum, Denmark)

A two-microphone sound intensity system is, in general, calibrated by using a pistonphone or a sound level calibrator to set the sensitivities of the two measurement channels. While this is all that is required for a calibration of the system, it does not provide any check of the systems performance. Such checks have traditionally been limited to the laboratory because they require special acoustic environments. A new device for calibrating and checking intensity systems will be considered here. This device consists of an acoustic source and a coupler. The coupler can be configured in two ways. In one configuration both microphones are exposed to the same sound field. This mode can be used to calibrate the channel sensitivities and to measure the residual intensity index of the system. In the other configuration the coupler will simulate free-field conditions. Either a pistonphone or a broadband source can be used as the acoustic source. The design of the device will be considered and its use illustrated.

11:00

**EE10. A multimicrophone system with spatial selectivity.** Hikaru Date (Department of Information Engineering, Yamagata University, 4-3-16, Johnan, Yonezawa, 992 Japan) and Ken'ichi Furuya (Acoustic Department, Kyushu Institute of Design, 4-9-1, Shiobaru, Minami, Fukuoka, 815 Japan)

The principle of spatial selectivity of a receiving system, which comprises several microphones distributed on a spherical boundary surface and one microphone at the center of the space enclosed by the boundary, is introduced in accordance with the theorem expressed by Kirchhoff's integral equation. The output from each of the boundary microphones are processed in the following way. Each signal is first delayed and then differentiated in both time and space in the direction normal to the spherical boundary. All of the delayed and differentiated signals are added together with appropriate weighting. The system output is finally obtained by subtracting the weighted sum from the output of the center microphone. The system output contains no components of external noise from the outside of the spherical boundary. Simulation studies were carried out for various numbers of microphones and for different ratios of signal wavelength to the radius of the spherical boundary and for both plane and spherical incident waves.

11:15

**EE11. Absolute calibration of acoustic sensors utilizing electromagnetic scattering from *in situ* particulate matter.** Allan D. Pierce and Yves H. Berthelot (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

The present paper initiates a study to understand the errors inherent to a laboratory technique similar to that used by K. J. Taylor [J. Acoust. Soc. Am. 70, 939-945 (1981)] for absolute calibration of microphones. Considered here is the idealized situation when a plane-polarized monochromatic plane electromagnetic wave is incident on a finite scattering volume containing dispersed small dielectric spheres that oscillate under the joint influence of a plane transient acoustic wave and Brownian motion. Any component of the scattered and Doppler-shifted electromagnetic signal at a distant farfield point in an obliquely backward direction is a sum of  $N$  terms, each of the generic form  $A_n \cos(\omega_0 t) + B_n \sin(\omega_0 t)$ , where  $A_n$  and  $B_n$  depend on the instantaneous position of the  $n$ th scatterer. Since these positions change with time, the coefficients also vary, although slowly over intervals comparable to  $1/\omega_0$ . Statistical properties of the time varying sum functions  $A(t)$  and  $B(t)$  are studied analytically and with numerical simulations for various models of the sound wave and the Brownian motion. A principal problem addressed is that of determining the statistics of the time interval  $\Delta t$  over which a fixed number  $K$  of zero crossings occur for the sum  $A(t) \cos(\omega_0 t) + B(t) \sin(\omega_0 t)$ , given that this interval is centered at a particular time, the pertinent quantity of interest being the relative error in the accumulative phase difference  $2\pi K / \Delta t - \omega_0$  per unit time. The results support the general conclusions that smaller errors are expected for higher amplitude sound waves and for sound waves of lower frequency. [Work supported by ONR.]

11:30

**EE12. Differences in measured microphone responses at different distances from artificial mouth.** Frederick M. Kruger (Kruger Associates, 37 Somerset Drive, Commack, NY 11725), Mitchell Mayer (Sonetronics, Inc., 1718 H Street, West Belmar, NJ 07719), John Bareham (Brüel & Kjær, Marlboro, MA 01752), and Barbara Kruger (Kruger Associates, 37 Somerset Drive, Commack, NY 11725)

When calibrated and operated at different distances, available artificial voice/mouth frequency response measuring systems yield significantly different results. These differences are greater for different distances than for different voice/mouths. Virtually all reported voice measurements are made at the CCITT recommended distance of 25 mm from the lips, or 25 mm in front of the "lip ring" of the artificial voice/mouths. Yet, all high-noise environment military microphones are tested 1/4 in. (6 mm) from the voice/mouth opening. This is, however, several millimeters behind the *lip plane* of the voices. At this distance, noise canceling microphones show minimal cancellation effects. Historically, the use of 1/4-in. distance began with a laboratory artificial voice/driver sound system, which predated the first commercially available, relatively large scale production artificial voice (B&K 4216). This paper shows the effect(s) on measured microphone response when military noise canceling microphones (M-138, H-250, M-162) and a calibrated laboratory microphone are tested, on axis, at different distances from several calibrated voices (6 mm from opening, 6 and 25 mm in front of lip ring). Real world implications are discussed.

**Session FF. Speech Communication VI: Topics in Speech Perception (Poster Session)**

Fred D. Minifie, Chairman

*Department of Speech and Hearing Sciences, University of Washington, JG-15, 1417 N.E. 42nd Street, Seattle, Washington 98195***Contributed Papers**

All posters will be displayed from 8:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

**FF1. Consonant and vowel intelligibility with the Symbion cochlear implant.** Michael F. Dorman (Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85287)

Studies of consonant intelligibility (16 items in aCa format) and vowel intelligibility (12 items in bVt format) were conducted with six patients fitted with the Symbion four-channel cochlear implant. Consonant intelligibility ranged from 52% to 81% correct. Voicing and manner errors were uncommon. Vowel intelligibility ranged from 22% to 79% correct. Errors were usually vowels with similar *F1* and/or *F2*. For the best patient, an analysis of errors as a function of channel stimulated indicated that 81% consonant intelligibility could be achieved with the use of two functional channels, one centered at 500 Hz and the other centered at 4000 Hz.

**FF2. Speech perception: The effects of recurrent conductive hearing loss in young children.** Richard L. Clarkson (Rhode Island School for the Deaf, Corliss Park, Providence, RI 02908), Peter D. Eimas (Department of Psychology, Brown University, Providence, RI 02906), and G. Cameron Marean (Department of Speech and Hearing Sciences, JG-15, University of Washington, Seattle, WA 98195)

The ability of five-year-old children with histories of recurrent otitis media (OM) and associated conductive hearing loss to perceive syllable initial stop consonants was investigated. Naturally produced, computer-edited stimuli were created to form continua that ranged in voice onset time (VOT) from 7 to 60 ms. The ability of children to assign these stimuli to adult phonetic categories was assessed by identification and discrimination tasks. Three subject groups were identified: One group (group C) had no history of OM and was linguistically age appropriate; another group (group OM) had considerable involvement with OM and was also linguistically normal; the third group (group OMDL) had an equal involvement as group OM with otitis media, but was linguistically delayed. All groups were matched for age, IQ, socioeconomic class, and educational experience. Results indicate that OM was associated with poorer discrimination scores, regardless of the linguistic status of the individual.

**FF3. Infant speech-sound discrimination in noise.** Robert J. Nozza, Linda C. Bond, Sandy L. Miller, and Reva N. F. Rossman (Division of Audiology, Children's Hospital of Pittsburgh, One Children's Place, Pittsburgh, PA 15213)

Infant ability to discriminate a speech-sound pair in noise was assessed using the visual reinforcement infant speech discrimination procedure. Sixteen infants between 7 and 12 months were tested on the /ba-ga/ contrast at four S/N's (-8, 0, 8, and 16 dB). A group of young-adult control subjects was tested also at four S/N's (-12, -8, -4, and 0 dB). The noise was continuously present at 48 dB SPL. Group psycho-

metric functions revealed that infants required a more favorable S/N than adults to achieve given levels of performance. The adult subjects and 14 of the 16 infants were also tested using a discrimination-threshold procedure in which the stimulus intensity was varied adaptively (1-up/1-down). The adaptive discrimination procedure provided, for both groups, estimates of performance that are in good agreement with those from the performance versus S/N functions that were developed using the more time-consuming method of trial blocks at fixed S/N's. [Work supported by DRF and NIH PO1 NS16337.]

**FF4. Relation of children's identification of consonant-vowel syllables to their fine-grained auditory discrimination: A progress report.** Lois L. Elliott, Michael A. Hammer, Margo E. Scholl, and Bonnie S. Anthony (Northwestern University, 2299 Sheridan Road, Evanston, IL 60208)

Just-noticeable differences (JNDs) and identification (i.e., labeling) functions were obtained for children aged 6 through 11 years. Two consonant-vowel (CV) continua representing the place-of-articulation and voice-onset-time features served as stimuli; the adaptive discrimination task that has been previously described [e.g., L. L. Elliott *et al.*, *J. Acoust. Soc. Am.* 70, 669 (1981)] was used. Some children produced consistent identification functions for both continua; some were inconsistent in labeling CVs of one continuum only; and some were inconsistent for both continua. In many cases, the children with the poorest identification performance also exhibited the poorest JNDs for the same stimulus continuum. The irregular identification functions observed for some of these normal-hearing children resembled functions that have been reported for some adults [B. E. Walden *et al.*, *J. Acoust. Soc. Am.* 79, 1101 (1986)]. [Work supported, in part, by NINCDS (NIH).]

**FF5. Normative data and physical measurements for a nonsense CVC intelligibility test.** Christine M. Rankovic and Dianne J. Van Tasell (Department of Communication Disorders, University of Minnesota, Minneapolis, MN 55455)

A nonsense CVC stimulus set has been developed for investigations into the application of the articulation index (AI) to speech recognition testing with hearing-impaired persons. Each test list consists of 57 items drawn at random from a corpus of 228 items comprising syllables spoken by four talkers. Crossover functions support the similarity of this stimulus set to that of French and Steinberg [*J. Acoust. Soc. Am.* 19, 90 (1947)] in terms of frequency importance function. Because the number of vowels was restricted to three, the long-term spectrum of the stimulus set is less smooth than that usually associated with connected speech. The AI versus percent-correct phoneme score functions have been obtained from normally hearing subjects. Comparisons with published data for similar stimulus sets will be made. Talker differences, reliability of subject perfor-

mance, and preliminary data from hearing-impaired subjects will be reported. [Work supported by NICHHD T32 HD-07151 and NINCDS NS 12125.]

**FF6. Patterns of phoneme confusions with sense and nonsense CVC words for subjects with normal hearing and with presbycusis.** Arjan Bosman and Guido F. Smoorenburg (Laboratory of Experimental Audiology, Department of Otorhinolaryngology, University Hospital, Catharijnesingel 101, Utrecht 3511 GV, The Netherlands)

Differences in listening strategy between normal-hearing and hearing-impaired listeners will become apparent in different patterns of phoneme confusions. The confusions were studied with meaningful and meaningless word material of the consonant-vowel-consonant type (CVC words) presented in quiet at fixed presentation levels to a group of young normal-hearing subjects, a group of elderly subjects with normal hearing regarding their age, and subjects with presbycusis. The confusions of the vowels and initial and final consonants were subjected to INDSCAL analysis [J. D. Carroll and J. J. Chang, *Psychometrika* **35**, 283-319 (1970)]. Parameters in the analysis were; word type (meaningful versus meaningless), group of subjects, and phoneme identification score. This latter parameter is perceptually more relevant than presentation level when comparing subjects with different hearing losses. For the normal-hearing subjects, vowel perception was dominated by the first and second formant, whereas for presbycusis subjects the contribution of the second formant was reduced and some influence of vowel duration was present. For the perception of both initial and final consonants voicing was all important. However, the feature of voicing was better perceived by presbycusis subjects than by the normal-hearing subjects.

**FF7. Relations between the intelligibility of sentences and words for subjects with normal hearing and with presbycusis.** Guido F. Smoorenburg and Arjan Bosman (Laboratory of Experimental Audiology, Department of Otorhinolaryngology, University Hospital, Catharijnesingel 101, Utrecht 3511 GV, The Netherlands)

In order to see whether hearing-impaired listeners make more extensive use of nonacoustic cues than listeners with normal hearing, different speech materials were presented to three groups of subjects. The speech material consisted of meaningful and meaningless words of the consonant-vowel-consonant type (CVC words) and short, simple sentences of eight to nine syllables [R. Plomp and A. M. Mimpen, *Audiology* **18**, 43-52 (1979)]. All material was uttered by a male and a female speaker. Three groups of subjects participated in the experiments: young normal-hearing subjects, elderly subjects with good hearing regarding their age, and subjects with presbycusis. Test-retest reliability for both word and sentence material was found to be close to the theoretical value. Averaged across all groups of subjects, the correlation between the SRT for word and the SRT for sentence material was 0.97, which resulted in a prediction error of 5.3 dB. Prediction of the SRT's from the tone audiogram with the Fletcher index (average hearing loss at 500, 1000, and 2000 Hz) resulted in a prediction error of 5.5 and 6.8 dB for the SRT for word and sentence material, respectively.

**FF8. Source characteristics of diplophonia.** Bruce R. Gerratt, Kristin Precoda, David G. Hanson, and Gerald S. Berke (Veterans Administration Medical Center, West Los Angeles, Wilshire and Sawtelle Boulevards, W126, Los Angeles, CA 90073 and UCLA School of Medicine, Los Angeles, CA 90024)

Diplophonia describes the perception of more than one fundamental frequency component in a voice. Simultaneous photoglottographic (PGG), electroglottographic (EGG), and acoustic signals were collected from five diplophonic subjects. Results indicate that diplophonia is similar to the beat frequency phenomenon caused by independent vibration of the two vocal folds at slightly different frequencies. In the time domain, modeling vocal fold displacement from midline by summing two sine waves produced area waveforms closely resembling the PGG wave-

forms obtained from the diplophonic subjects. In the frequency domain, the diplophonic voice spectrum bears little resemblance to that of normal phonation. Instead of the normal set of harmonic peaks at integer multiples of the fundamental, the diplophonic spectrum shows a wealth of very closely spaced energy peaks of which the two fundamentals are often the most prominent but rarely the lowest frequency. The spectral envelope of the glottal area waveform may appear to have poles where no formants are possible and the spectral energy falls off quite rapidly. The locations of prominent energy peaks were predicted by modeling the displacement of the vocal folds from midline using the sum of two sine waves clipped by pulse trains to simulate collision effects of the two vocal folds. [Work supported by NINCDS.]

**FF9. Elicitation of spontaneous speech production from preschool-aged tactile aid users.** Adele Proctor (Northeastern University, Boston, MA 02115) and Corine Bickley (Massachusetts Institute of Technology, Cambridge, MA 02139)

Engineering of sensory supplements that are sufficiently small in size, light in weight, and adequate protocols for measuring device efficacy have long been problems for those involved in designing vibrotactile aids for preschool-aged deaf children. Contributing to the engineering variables influencing tactile aid development are the problems of how to elicit representative samples of spontaneous speech production before, during, and after tactile aid usage and how to quantify effects of tactile aid usage on speech production of children who exhibit nonnormal articulatory patterns. Since visual displays presented on microprocessors are reported to be effective elicitors of vocal output, available databases on articulatory and acoustic characteristics of hearing and hearing-impaired preschoolers, primary acoustic parameters known to influence intelligibility, and available data suggesting effects of tactile aids on speech production were incorporated to design graphic stimuli to elicit speech from 3- to 5-year-old deaf tactile aid users. The purpose of this poster session is to discuss the theoretical basis for selection of stimuli, to demonstrate the protocol for eliciting spontaneous productions, and to present the planned articulatory and acoustic measurements. [Work supported by DOE, Grant No. 133 GH70189.]

**FF10. Acoustic characteristics of dysarthria associated with cerebral palsy.** Gary Weismer and Karen Forrest (Speech Motor Control Laboratories, University of Wisconsin—Madison, Madison, WI 53705)

This paper will report on qualitative and quantitative observations on the speech production characteristics of adults with cerebral palsy. The subjects were ten adults with cerebral palsy, primarily of the spastic type. Multiple repetitions of selected sentences were analyzed acoustically to determine the spectral and temporal characteristics of segments. In addition, certain qualitative analyses, involving the form of spectrographic displays across utterance repetitions, were performed to address the issue of random versus consistent control of articulatory gestures in neurogenic speech disorders. It is argued that in some cases a patient may produce very aberrant temporal and spectral features which, nevertheless, are quite consistent across repetitions.

**FF11. Acoustic characteristics of speech produced by an older geriatric population.** Julie M. Liss, Gary Weismer (Department of Communication Disorders, 1975 Willow Drive, University of Wisconsin, Madison, WI 53706), and John C. Rosenbek (Middleton Memorial Veterans Administration Hospital, Madison, WI 53705)

The effect of advancing age on speech production characteristics is not clearly understood. Yet it is crucial for those who study and serve the normal and neurologically impaired geriatric populations. The purpose of this study was to obtain a normative database for the speech production characteristics of an older geriatric group. Fourteen veterans, 87 to 93 years old, served as subjects. A total of 40 sentences were produced at a conversational rate by each gentleman. Wideband (300-Hz) spectrograms were created from high-quality tape recordings. Specified acoustic

measures were made (consonant, vowel, and voice onset time durations, and vowel formant trajectories) via a bit-pad digitizer and microcomputer sonogram analyzer program. Descriptive analysis was performed to obtain normal parameters of speech production. Current theories of aging and neurological diseases affecting speech will be discussed.

**FF12. Analysis of formant transitions for English fricatives spoken by five male and five female talkers.** Kelsey Key and Dennis H. Klatt (Room 36-523, Massachusetts Institute of Technology, Cambridge, MA 02139)

Nonsense words of the form C'VC'VC'VC were recorded for the four voiceless fricatives of English [f,th,s,sh] in ten vowel environments by each of five male and five female talkers. The database was digitized at 16 000 samples/s and digital spectrograms were produced. Formant frequency values,  $F_2$  and  $F_3$ , were measured by hand from the spectrograms at voicing onset and vowel midpoint for each CV transition. These data were examined in order to seek regularities of several possible forms: (1) invariant onset frequencies independent of speaker and vowel environment, (2) existence of locus frequencies that differ as a function of place of fricative articulation, but are similar across speakers, and (3) similar transition behavior (e.g., rising or falling  $F_2$  and/or  $F_3$ ) across speakers for a given vowel with no attempt to seek regularities across vowels. In all cases, distributions of transition-related measures across speakers for each place of articulation show disturbing amounts of overlap, even when male and female data are plotted separately. While it has often been said that [f] and [th] are distinguished perceptually by  $F_2$  and  $F_3$  transition behavior (because frication spectra are often quite similar), the data suggest that it will be difficult for a speech recognition device to make this distinction reliably from observed formant transition behavior in the general population. Some reassessment of the appropriate perceptual cues seems to be required. [Work supported in part by an NIH grant.]

**FF13. Acoustic properties of isolated stop consonant release transients.** Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695) and Hwei-Bing Lin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695 and Department of Linguistics, University of Connecticut, Storrs, CT 06268)

This study focuses on the initial component of the stop consonant release burst, the release transient. In theory, the transient, because of its impulselike source, should contain much information about the vocal tract configuration at release, but it is usually weak in intensity and difficult to isolate from the accompanying frication in natural speech. For this investigation, a human talker produced isolated release transients of /b,d,g/ in nine vocalic contexts by "mouthing" these syllables. He also produced the corresponding CV syllables with regular phonation for comparison. Spectral analyses showed the isolated transients to have a clearly defined formant structure, which was not seen in natural release bursts whose spectrum was dominated by the frication noise. The formant frequencies varied systematically with both consonant place of articulation and vocalic context, with the latter, "coarticulatory" influence actually exceeding the former. The transient thus provides an acoustic snapshot of the vocal tract immediately after the release. [Work supported by NICHD.]

**FF14. Some phonetic characteristics of sentences produced in noise.** Z. S. Bond (Department of Linguistics, Ohio University, Athens, OH 45701), Thomas J. Moore (Armstrong Aerospace Medical Research Laboratory, Wright-Patterson AFB, OH 45433), and Kate McCreight (Department of Linguistics, Massachusetts Institute of Technology, Cambridge, MA 02139)

In previous work, relatively systematic differences were found in the phonetic structure of spondee words produced in isolation and in the presence of noise. The purpose of this study was to determine whether noise affects continuous speech in a similar manner. Four speakers were recorded reading 20 sentences, two times in quiet and two times while

hearing 95 dB SPL pink noise. The sentences were digitized, segmented, and transcribed using SPIRE. The resulting database consisted of approximately 850 segments per subject per speaking condition. The distributions of spectral and temporal properties of classes of segments were determined using SEARCH. All subjects produced sentences in the presence of noise with increases in fundamental frequency and total energy, as has been found for isolated words. Segment durations and spectral characteristics were affected by noise for some subjects more than for others. [Work supported in part by AFOSR.]

**FF15. Fundamental frequency contours of preword vocalizations through the emergence of word combinations.** Michael P. Robb (Department of Speech Pathology and Audiology, University of Hawaii, Honolulu, HI 96822) and John H. Saxman (Communication Sciences and Disorders Program, Syracuse University, Syracuse, NY 13244)

An increasing number of acoustic characteristics of infant preword vocalizations have been found to be similar in many respects to the vocalization characteristics of children's early meaningful speech. Such findings are suggestive of continuity in the underlying processes involved in acquiring language. To date, few studies exist that have examined in detail the acoustic characteristics of babbling, early words, and word combinations in a single group of children. The present study provides monthly data on fundamental frequency ( $F_0$ ) contours in seven infants who were audio tape recorded over a 12-month period. Within a year's time, each child's communication development progressed from babbling vocalizations to word combinations. The frequency and type of  $F_0$  contours in each child's mono-, bi-, and polysyllabic vocalizations are reported. Regardless of syllable type, the children used a large number of rising-falling  $F_0$  contours across preword and multiword periods of vocalization development. The least frequently occurring  $F_0$  contours in both group and individual data were falling-rising contours. It is proposed that uniformity exists in early vocalizations across preword and meaningful speech periods due to physiologically motivated factors.

**FF16. Classification of vowels using spectrum moments.** Paul Milenkovic and Karen Forrest (Waisman Center, University of Wisconsin, 1500 Highland Avenue, Madison, WI 53705-2280)

Formant frequencies form a set of acoustic features that can be used to classify vowels phonetically [A. K. Syrdal and H. S. Gopal, *J. Acoust. Soc. Am.* 79, 1086-1100 (1986)]. The advantage of formant frequencies is that they are unaffected by changes in the speech spectrum slope attributable to the voice source. The disadvantage of formant frequencies is in the large errors in frequency that can result from a misclassification of the formant peaks of the speech spectrum. A set of features based on statistical moments computed from the speech spectrum has been successful in classifying a broad class of obstruent sounds [K. Forrest *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 82, S84 (1987)]. In the case of vowel sounds, moment analysis eliminates the problem of identifying formant peaks but introduces the problem of variability in the moment features resulting from changes in the voice source. In order to correct for between subject variation in the voice source, the long-term average spectrum of all of the speech samples recorded from each subject is computed and used to normalize vowel spectra prior to computing moments. Between subject classification results for adult subjects are presented. [Work supported by NIH grants NS 21516 and NS 13274.]

**FF17. Factor analysis for vowel and fricative duration in American English.** John F. Pitrelli and Victor W. Zue (Room 36-575, Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology, Cambridge, MA 02139)

In an attempt to develop a comprehensive model of segment duration for American English, the duration of vowels and fricatives as a function of segmental and suprasegmental factors was studied. Two databases were used, one consisting of one reading of the 720 Harvard List sentences by one male speaker, and another consisting of the first 500 Harvard List



sentences, with each list of ten read by one male and one female speaker. Segment durations were obtained from time-aligned phonemic transcriptions. Factors influencing duration and their interaction effects were studied using analysis-of-variance techniques. Results indicate that vowel durations are largely affected by factors such as vowel class (diphthongs, tense, lax, and reduced vowels), sentence-final lengthening, the syntactic function of the word in which they appear, and lexical stress. Other factors previously thought to be important, such as the voicing characteristic of the following consonant, did not account for much of the variance in the data. Using five of the most significant factors, an account was obtained for 67% of the variance on training data as well as on test data for the single-speaker database, and 63% and 59% for the multispeaker database. The difference in results for the two databases may be attributable to interspeaker differences that include speaking-rate variations. In an oral presentation, the results on the modeling of fricative duration will be discussed. [Work supported by DARPA under contract N00014-82-K-0727, monitored through the Office of Naval Research.]

**FF18. Spectrographic comparison of whispered voiced and voiceless stop consonants in various vowel environments.** Igor V. Nábělek, Hsiao-Chuan Chen (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 36996-0740), and Sumalai Maroonroge (Speech and Hearing Clinic, Ramathibodi Hospital, Bangkok 10100, Thailand)

A previous presentation reported on spectra of whispered and phonated speech and on comparison of identification of whispered voiced and voiceless consonants when those consonants were presented in C/a/ syllables. The cues for voicing in whispered plosives were studied using spectrograms. In a follow-up study, spectrographic analysis of whispered monosyllables with plosives in various vowel environments was performed. The results of the analysis—with respect to the voicing differences—will be presented and discussed.

**FF19. Voice "features" in long-term memory.** Jody Kreiman (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024) and George Papcun (Los Alamos National Laboratory, Computer User Services C-10, Mail Stop B-296, Los Alamos, NM 87545)

This study examines the perceptual parameters used to recognize an unfamiliar voice (heard only once before) after a one-week delay, as compared to the parameters used to discriminate among the same voices in a short-term memory (paired comparison) task. Ten groups of ten listeners each heard a short sample of a single voice; after seven days, all listeners heard the full ten voice set and had to determine if their target voice was present or not. Confusion data and similarity ratings were gathered and combined to form full matrices that were analyzed using multidimensional scaling. Differences in the perceptual parameters that underlie confusions and perceived similarities in the short- versus the long-term memory tasks will be discussed, as will changes in the relative importance of different parameters with time. The implications of these findings for models of voice perception and speaker recognition will also be addressed.

**FF20. A neuropsychological model of voice perception.** Diana Van Lancker (Psychiatry Department, UCLA, Los Angeles, CA 90024) and Jody Kreiman (Phonetics Laboratory, Linguistics Department, UCLA, Los Angeles, CA 90024)

It will be argued that two neuropsychological processes mediate voice perception: pattern recognition and featural analysis. Properties of voices that determine the relative dominance of these two kinds of processing include: (1) the familiarity of the voices, and (2) the heterogeneity of the voice set—how "similar" the voices are to each other. Familiar voices that differ substantially are processed largely as complex patterns, while a homogeneous set of unfamiliar voices recruits some featural analysis. This model is derived from the following observations: (1) Acoustically altering familiar voices affects their recognizability unevenly, suggesting that no fixed set of parameters underlies the recognition process; (2) perfor-

mance of dichotic listening and brain-damaged subjects indicates that the right hemisphere is specialized for recognizing familiar voices (the right hemisphere is believed to be superior in pattern recognition); (3) in paired comparison tasks, normal subjects compare discrete auditory features, as well as the overall pattern; (4) brain-damaged subjects who show impaired performance in unfamiliar voice discrimination have damage to either the right or left temporal lobe (specialized for processing of auditory detail). Testable hypotheses generated by this model will also be discussed.

**FF21. Speech rate and gender as mediating factors in person perception.** Faith-Anne Dohm, Stanley Feldstein, and Cynthia L. Crown (Department of Psychology, University of Maryland, Baltimore County, 5401 Wilkens Avenue, Baltimore, MD 21228)

A previous study by the authors found that judgment of speech rate is independently influenced by the gender of the judges and of the speakers. The research of others [e.g., B. Brown *et al.*, *J. Acoust. Soc. Am.* 54, 29–35 (1973)] has shown that speech-rate perception mediates other aspects of interpersonal perception. The study reported here, then, investigated the possibilities that: (a) Individuals who speak rapidly are viewed in more positive ways than those who speak slowly; (b) males are viewed in more positive ways than females; and/or (c) speech and gender jointly influence person perception. Seventeen male and 28 female college students listened to and rated, each of ten adjective scales, three male and three female speech samples. An hierarchical multiple regression analysis indicated that, whereas the male speakers received less positive ratings when they spoke rapidly, the female speakers received less positive ratings *both* when they spoke rapidly and slowly. The results suggest that person perception is influenced jointly by the speech rate and gender of a speaker.

**FF22. Turning the teaching of foreign pronunciations into a computer product.** Raymond J. Thomas (Faculté des Sciences de Luminy, 13288 Marseille Cedex 9, France)

Extending the scientific teaching of English and French pronunciations [*J. Acoust. Soc. Am. Suppl.* 1 76, S90 (1984)] to adults speaking other languages showed that specific variations in speech acoustics and use of muscles are related basically in the same manner in all humans, but that an added complexity arises from seemingly unrelated speech habits depending on each mother tongue. Final results of advice are thus unpredictable, but practice has shown that progress on various points could be totaled through gradual classifications of oppositions. This could be achieved with a computer program using the speech syllabic data bank (Proc. XIth ICPHS, Tallinn, 1987, Sec. 89.4) together with a much smaller bank on rhythms and stresses in the various languages. The study of fewer than 20 phonetic systems would cover all important speech habits. This would result in the capacity for the computers to "think out," in a matter of minutes, the one among the 16 000 000 odd teaching programs it could devise (Evaluation de l'information en langues naturelles, Ms de Thèse, BU de Rouen, 1983, Vol. 3) that would be best adapted to anyone of us for learning any of the 4000 odd foreign pronunciations.

**FF23. Time-gated word identification performance.** Chie Higuchi Craig (Department of Speech Pathology and Audiology, University of Wisconsin-Milwaukee, Milwaukee, WI 53201)

Listeners who are able to identify words from their earliest onset demonstrate an effective integration of acoustic-phonetic and predictive-contextual cues for real-time speech recognition. In this investigation, a gating paradigm was used to study how presentation time (duration from word onset) influences word identification performance. A digital sonagraph was programmed to store and time gate 50 isolated monosyllabic words drawn from a recording (Auditec) of the NU-6 (list 1A). Twenty normal-hearing young adults were tested monaurally via earphones. Each word was presented repeatedly with presentation times increasing in 60-ms increments with each successive pass. Words were presented in 3–7 passes until each word was completely presented (180–420 ms). Subjects

were asked to guess the word, write it down, and indicate how confident they were about each guess. Results reveal the effect of a 60-ms gating paradigm on word identification performance and relations between word duration, listener confidence, and percent of words identified.

**FF24. Spectral shaping and the intelligibility of speech in noise.** R. Pedlow (SRL/Psychology Department, Indiana University, Bloomington, IN 47405)

Various studies have shown that attenuation of the first formant results in enhanced speech intelligibility in noise. One explanation (Nye *et al.*, 1974) suggests that it is caused by upward spread of masking in normal speech perception. The other explanation proposed is that attenuating  $F_1$  increases the amount of intelligibility information relative to the total speech energy in the signal. The stimulus materials were phonetically balanced (PB) words produced by one male speaker. Stimuli were digitally filtered with a high-pass filter that had a passband from 1100–5000 Hz and a stopband from 0–850 Hz (the attenuation levels of the stopband were 10, 20, 40, and 80 dB relative to the unfiltered signal). Items were equated for rms energy after filtering. Processed stimuli were presented to subjects in a perceptual identification task at 0 dB S/N. Articulation index values were then computed for the test items. Fitting transfer functions to the data showed that the increased intelligibility obtained by high-pass filtering was not well predicted by the articulation index values. This supports the hypothesis that attenuating the  $F_1$  results in an increase in intelligibility due to a release from masking effect.

**FF25. Perceptual evidence for stress foot structure.** Jan Edwards (Hunter College, 425 East 25 Street, New York, NY 10010) and Mary Beckman (Department of Linguistics, The Ohio State University, Columbus, OH 43210)

A year ago, Beckman and Edwards [J. Acoust. Soc. Am. Suppl. 1 81, S67 (1987)] reported on two durational effects. Phrase-final lengthening is a phonologic mark of intonational phrase boundaries: It is a substantial effect whose domain is highly consistent across speakers. Seeming word-final lengthening is a much smaller and more variable effect—how to describe the domain of this smaller effect, or even whether it should be described phonologically, was unknown. The perceptual salience of these two effects has been examined, reasoning that if they are phonologic marks of prosodic constituent boundaries, then listeners should be able to perceive the edges so marked. Listeners can consistently use intonational phrase-final lengthening, but not the smaller durational effect, to discriminate between phrases that are otherwise identical. It is suggested that the smaller effect is not a phonologic mark of a prosodic constituent within the intonational phrase. Rather, it is a result of articulatory timing relationships within the stress foot, with the longer [pa] in 'pop opposed' as compared to 'poppa posed' due to their different stress feet structures. [Work supported by NSF Grants. No. IRI-8617873 and IRI-8617852.]

**FF26. Do segmental attributes distinguish spontaneous and prepared speech?** Robert E. Remez, Anne Sherman, Thea S. Klapwald (Department of Psychology, Barnard College, 3009 Broadway, New York, NY 10027), and Philip E. Rubin (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

A set of perceptual studies was conducted to assess the salience of segmental phonetic differences between spontaneous and prepared speech. Prior studies employing our perceptual paradigm revealed that naive listeners can differentiate lexically, syntactically, and thematically identical sentence pairs that have been extracted from brief monologs (> 40 s), identifying the spontaneously uttered version at high levels of performance. No single acoustic attribute appears to predict the perceptual salience of spontaneity in production, though the variability of  $f_0$  as the sentence ends and the average duration of syllables are both correlated with the detection of spontaneity. The present investigation exploits the perceptual differentiation test to determine the salience of segmental differences between spontaneous and prepared utterances. Against the

benchmark of differentiability of intact sentence pairs, the perceptual distinctiveness of spontaneous and read utterances was evaluated by deleting the segmental information alone by low-pass filtering. In additional converging tests, versions of the sentence pairs without formant variation were synthesized, setting the spectrum to the vowel [o], while preserving natural differences in duration and  $f_0$ ; and, synthetic sentence pairs lacking variation in  $f_0$  were tested in filtered and [o] versions. The results will be discussed in reference to the perceptual constraints on the comprehension of spontaneous speech. [Research supported by grants from NINCDS and NICHD.]

**FF27. Learning effects for two different speech synthesizers with different speech material.** Rory A. DePaolis (The Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802)

This study examined the relationship between the learning effects due to two inexpensive methods of synthesizing speech, formant synthesis (FS) and linear predictive coding (LPC). An experiment using two groups of six subjects listening to either FS or LPC for four consecutive days was conducted. The test material consisted of contextual and nonsense sentences with written clues, contextual sentences without written clues, and the rhyme test. A prose passage was also presented to the subjects. There was no significant difference between the learning curves for the two synthesizers but there was an overall dramatic learning effect, most noticeably from the first to third presentation across all types of speech material. The synthesizers exhibited equality of mean percent correct word scores for all material over the four sessions, excluding day 1, using the nonsense sentences in which the FS device was more understandable. These results are presented graphically with comparisons to other studies using natural and synthetic speech and predictions on the performance of the two synthesizers in noise.

**FF28. Perception of digitally coded speech under some cognitive load.** Kazunori Ozawa (C&C Information Technology Research Laboratories, NEC Corporation, 4-1-1, Miyazaki, Miyamae-ku, Kawasaki 213, Japan) and John S. Logan (Speech Research Laboratory, Department of Psychology, Indiana University, Bloomington, IN 47405)

Perception of meaningful sentences that were encoded by speech coding methods was measured. In order to differentiate and to expand perceptual differences between the speech coding methods for the meaningful sentences, a task of recall for the sentences under cognitive load was adopted. In this experiment, the distractor method of short-term memory [L. R. Peterson and M. J. Peterson, J. Exp. Psychol. 58, 193–198 (1959)] was utilized as the cognitive load to subjects. Pitch predictive multipulse speech coding at 8 kb/s [K. Ozawa and T. Araseki, Proc. ICASSP, 1689–1692 (1986)] and 50 kb/s  $\mu$ -law PCM were used as speech coding methods. As a reference, uncoded natural speech was also used. A total of 60 Harvard sentences was used as meaningful sentence stimuli. Results showed that the higher the cognitive load, the larger the perceptual differences between uncoded and coded speech, as well as between MPC and PCM. However, differences in error rates among uncoded and coded speech were very small in the case of no cognitive load. The results suggest that this perceptual evaluation method under cognitive load may be a sensitive method that can differentiate perceptual differences in speech quality between coded speech, even for meaningful sentences.

**FF29. Perception of /s/-consonant clusters.** Anna K. Nábělek, Tomasz Letowski, and Frances Tucker (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37996-0740)

Nonsense syllables /s/C/at/, where C was /p,t,k,f,m,n,l,w/, were recorded then degraded either by noise or by reverberation. The goal was to compare the effects of reverberant overlap of high-frequency energy of /s/ with masking by speech spectrum noise on the identification of the consonants. The results for 10 normal-hearing subjects indicated that the identification of consonants could be degraded by both noise and reverberation. In noise there was a predominance of manner errors whereas in

reverberation there was a predominance of place errors. It is possible that the place errors in reverberation were not related to masking by overlapping energy from /s/ but to temporal smearing within the consonants, which obscured the offset of the consonants. To assess if any masking by energy from /s/ could degrade identification of following consonants, the same syllables were tested with a white noise shaped with spectrum of /s/. In this condition, the consonants were highly identifiable. While energy in the consonant /s/ was relatively high, there was little overlapping of the high-frequency spectrum of /s/ with the low-frequency spectra of tested consonants. It is speculated that degrading effects of reverberation were related to the internal smearing within the consonants. [Work supported by NIH.]

**FF30. The role of formant relationships in the pattern processing of stop consonants.** Laurie F. Garrison (Department of Psychology, SUNY at Buffalo, Amherst, NY 14260)

Interaural transfer in selective adaptation was used to explore pattern processing in stop consonants. Specifically, the role of vowel category as proposed by Klatt [in *Perception and Production of Fluent Speech*, edited by R. A. Cole (Erlbaum, Hillsdale, NJ, 1980)] was investigated. Here, the onset characteristics of different consonants are "attached" to different vowel categories consisting of front, back, rounded, and unrounded vowels. This experiment tests for pattern processing based on these assumptions. The relationships between formants as well as vowel category for CV syllables were systematically manipulated. Five adaptors were constructed by changing the relationships between  $F_1$  and  $F_2$ . Two adaptors shared the same front vowel percept [bae]. Two other front vowel syllables, [bE] and [bI], and a back vowel syllable, [ba], were also used. These adaptors were then used in an interaural transfer adaptation experiment on a [bae-dae] series. The results will be discussed in relation to how changes in formant relationships and different vowel categories affect pattern processing of speech syllables. [Work supported by NINCDS.]

**FF31. Duration effects on vowel perception.** J. Besing, M. J. Collins, and J. Cullen (Division of Communication Disorders, Louisiana State University, Baton Rouge, LA 70803)

The effect of temporal differences on the identification of vowel tokens was studied using ten normal-hearing subjects. Stimuli were synthetically generated and varied in duration of the steady state, durations of the initial and final transitions, and  $F_1$ ,  $F_2$ , and  $F_3$  location. Listeners were required to identify the given vowel token from a set of ten possible alternatives. The results supported the established effects of duration on the perception of vowels; however, the duration effect was sensitive to the effects of initial and final transition duration and  $F_2$  location. These results indicate that alterations in the duration of the initial transition has greater impact than the final transitions. These effects as well as the effects of interactions between steady-state duration, initial and final transition duration, and  $F_1$ ,  $F_2$ ,  $F_3$  location will be discussed. [Research supported by NIMH.]

**FF32. Sensory-perceptual transformations in speech analysis.** James D. Miller, Steven J. Sadoff, and Mark R. Veksler (Central Institute for the Deaf, 818 S. Euclid Avenue, St. Louis, MO 63110)

In Miller's auditory-perceptual theory of phonetic recognition, the sensory effects of sequences of glottal-source spectra, burst-friction spectra, and silences are integrated into a unitary perceptual response by the sensory-perceptual transformations. The mathematical and computational implementations of the hypothesized transformations will be described. One transformation is applied to spectral pattern or shape and is implemented formant by formant. For example, the center frequency of a perceptual formant is calculated by second-order difference equations from the center frequencies, loudness, and bandwidths of the corresponding sensory formants of glottal-source and burst-friction spectra, when either or both are present, and from similar values for a neutral vocal tract during silence. The loudness of the perceptual response, which is calculated as the integrated loudness of the burst-friction and glottal-source inputs, decays slowly after the cessation of sensory input and, in this way, an

audible perceptual response can be maintained during brief periods of silence. Examples of application of these concepts to a variety of speech sounds, such as stops and fricatives, will be illustrated. [Work supported by NINCDS and AFOSR.]

**FF33. Coding of articulatory-acoustic events in a model of the peripheral auditory system.** Z. L. Wu, P. Escudier, J. L. Schwartz, and R. Sock (Institut de la Communication Parlée, U.A. CNRS INPG, 46 Av. Félix Viallet, 38031 Grenoble Cédex, France)

In the last years, several investigators have been concerned with speech processing in auditory models. These lines of research were carefully followed and research was restricted to auditory detection of a set of articulatory and acoustic events. The strategy consists of exploring various processing abilities lying in the model of the peripheral auditory system developed (see Dolmazon and Boulogne, *Speech Commun.* 1, 55-73), and attempting to demonstrate a possible adequacy between some of the physiological abilities and the acoustical structure of speech induced by articulatory constraints. Here, first results about characterization of voicing onsets and offsets and frication onsets and offsets for CVCV utterances in French will be presented and results obtained by various combinations of physiological mechanisms will be compared: with and without neural adaptation, with enhancement or diminution of low-frequency characteristics by temporally or geographically based filtering processes.

**FF34. Enhanced amplitude of the first harmonic as a correlate of voicelessness in aspirated consonants.** Carol Chapin Ringo (Research Laboratory of Electronics, Room 36-511, Massachusetts Institute of Technology, Cambridge, MA 02139)

Acoustic analysis was undertaken to test the hypothesis that information signaling voicelessness of a preceding aspirated consonant may reside in spectral characteristics associated with "breathy voice" at the onset of the following vowel. The study focused upon the enhancement of the first harmonic ( $H_1$ ) resulting from an increase in the open quotient of the voicing source waveform. The relative amplitudes of  $H_1$  and  $H_2$  were measured in vowels in CVd syllables pairing the consonants /p,t,k,h,b,d,g/ with the vowels /i,a,u,e,ɛ,ʌ/. The syllables were spoken by three male native speakers of English. Harmonic amplitudes were measured from DFT spectra (computed without preemphasis) of the first few pitch periods of the vowel. In general, the amplitude of  $H_1$  was greater than that of  $H_2$  for vowels following voiceless consonants; the converse was true for voiced consonants. A perceptual study using synthetic continua (varying in VOT) is being conducted; the continua differ in relative amplitude of  $H_1$  and  $H_2$ . Preliminary results indicate that for some subjects an enhanced  $H_1$  may contribute to a voiceless percept: The boundary between voiced and voiceless stops was shifted to smaller VOT values for stimuli in which the amplitude of  $H_1$  was greater than that of  $H_2$  at vowel onset. [Work supported by NIH.]

**FF35. An information-theoretic analysis of acoustic-phonetic and perceptual features.** Rick L. Jenison (Project Phoenix of Madison, Inc., 2001 S. Stoughton Road, Madison, WI 53716)

An information-theoretic approach using vector quantization (VQ) is presented as a framework for quantifying acoustic-phonetic distortion. An optimal codebook derived from 21 CV monosyllables from two speakers was generated for each of a contrasting set of linguistic features. The set of feature-based codebooks served as distortion minimized templates by which an average distortion could be computed for contrasting feature CV databases in quiet and in noise. The discriminating power of the feature codebooks based on a mean-square-error distortion was approximately 90 percent. Consonant confusion matrices were obtained from the CV database in quiet and in noise for five subjects. Transmitted information was computed for the same linguistic features analyzed by vector quantization. Preliminary evaluation of feature transmitted information compared to computed VQ distortion indicates a strong covariance. Further results and implications for distinctive feature systems will be discussed.

## Session GG. Physical Acoustics V and Bioresponse to Vibration II: Extracorporeal Shock Wave Lithotripsy—Physical Aspects

Lawrence A. Crum, Chairman

*National Center for Physics Acoustics, P.O. Box 847, University, Mississippi 38677*

Chairman's Introduction—8:25

### *Invited Papers*

8:30

**GG1. Physical aspects of lithotripsy.** David T. Blackstock (Applied Research Laboratories and Mechanical Engineering Department, University of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029 and Rochester Center for Biomedical Ultrasound, University of Rochester, Rochester, NY 14627)

Lithotripsy is a medical procedure by which focused shock waves are used to break up kidney stones. In the most widely used device, the Dornier lithotripter, the shock wave is produced by an electric spark at the near focus of an ellipsoidal reflector, which is located in a water bath. The patient is positioned so that the kidney stone lies at the far focus of the ellipsoid. The pressure signal there is a positive spike (headed by a shock) of amplitude as high as 1300 atm and duration about 1  $\mu$ s, followed by a negative pressure of longer duration and magnitude of order 100 atm. Particular emphasis in this talk is on the focused sound field, which is made up of the converging wave reflected from the ellipsoidal surface and a diffracted wave generated at the edge of the reflector. The changing relation between the two waves along the axis of the ellipsoid gives the lithotripter pulse its characteristic waveform. Also discussed briefly are the spark source, pressure signal measurement, cavitation, and stone breakup. [Work supported in part by NIH.]

9:05

**GG2. A comparison of commercial extracorporeal shock wave lithotripters based on measurements in the acoustic field.** Andrew J. Coleman and John E. Saunders (Medical Physics Department, Saint Thomas' Hospital, London SE1 7EH, England)

A survey of five, presently available, commercial extracorporeal shock wave lithotripters is reported. As part of the survey, a detailed examination of the acoustic field of the Dornier HM3, Wolf Piezolith 2200, Technomed Sonolith 2000, EDAP LT-01, and Siemens Lithostar was carried out using the same 25- $\mu$ m, bilaminar PVDF membrane hydrophone. The absolute pressure has been estimated from the voltage waveforms (recorded on a digital scope with a 30-MHz single-shot bandwidth) using the 1-MHz calibration factor for the hydrophone. The peak positive focal pressure at clinically used power settings of the different lithotripters is found to vary in the range 20–115 MPa. The peak negative focal pressure ranges from 3–10 MPa. The width of the focal region (defined as the region in which the peak positive pressure may be greater than 50% of that at the focus) varies from 0.1–2.5 cm. The rise time of the main pressure peak (defined as the time from 10%–90% of the peak positive pressure) is found to vary from less than 30 ns to 500 ns and the full width at a half-maximum of the positive pressure half-cycle from 250–500 ns. The estimated peak positive pressure gain of the different systems range from 10 to above 1000. Some of the parameters reported can be related to the relative clinical performance of the lithotripters. [Thanks to the companies and hospitals who provided the facilities for this survey.]

9:40

**GG3. High-intensity, focused ultrasonic fields.** Leif Bjørnø (Industrial Acoustics Laboratory, Technical University of Denmark, Building 425, DK-2800 Lyngby, Denmark)

The use of extracorporeal shock wave lithotripsy (ESWL) for disintegration of body stones has increased considerably during recent years. A worldwide activity in this field is reflected in a growing number of international publications and in the development and manufacturing of several ESWL machines marketed by companies in Germany and France, in particular. Two main types of ESWL systems are prevailing, the spark gap-based and the piezoelectric disk-based systems. This paper is introduced by a brief reconsideration of the features of pressure waves in water produced by an electrical discharge across a spark gap emphasizing the parameters of significance to pressure wave amplitude and time course. Moreover, a comparison is presented of results obtained using various theoretical models—including the KZK equation—for the calculation of pressure waveform parameters, spatial pressure distribution, etc. involving nonlinearity, diffraction, and absorption in the high-intensity focused ultrasonic fields produced by an ellipsoid as well as a spherical cap focusing

geometry. Data from the development of an ESWL of the piezoelectric disk type are reported including demands to transducers and to the electrical discharge circuit. Measured pressure waves in various regions of propagation will be presented. Calibration demands to transducers used for finite-amplitude pressure wave detection including linearity, dynamic range, etc. are emphasized. Finally, a critical assessment is given of the most essential pressure wave parameters, their relation to the focal system data, and their significance to stone disintegration efficiency.

10:15

**GG4. Extracorporeal shock wave lithotripsy: Method, application, and physical foundation.** Michael Grünewald, Heribert Koch, and Hajo Hermeking (Dornier Modizintechnik, Industriestr. 15, D-8034 Germering, West Germany)

During the last years a new device has been put into the medical health system allowing for noninvasive treatment of kidney and gall stones: the Dornier lithotripter. The principle methods of shock wave generation are discussed, such as underwater spark discharge, dielectric laser breakdown, electromagnetic and piezoelectric method, and their suitability to disintegrate body concretions effectively. For practical applications, careful choice of system parameters has to be made to fulfill all requirements: good stone disintegration and low pain and side effects for the patients simultaneously. Based on the theory of focusing the relevant physical effects like diffraction, nonlinear propagation, and attenuation (absorption) are explored both qualitatively and quantitatively, so far as possible. It will be shown that a full time-dependent profile calculation can be given within simple diffraction theory accounting for a proper description of the dependence of focusing on aperture, including stress and tensile. The relevance of pulse shape forming will be explored. Additionally nonlinear propagation in relation to damping and pulse deformation in soft tissue is important. Within a simple model the propagation of highly unharmonic pulses will be calculated including nonlinear steepening, anomalous tissue absorption, and dispersion.

10:50

**GG5. Cavitation microjets as a contributing mechanism for stone disintegration in ESWL.** Lawrence A. Crum (National Center for Physical Acoustics, P.O. Box 847, University, MS 38677)

The intense shock waves that are used for stone disintegration in ESWL can readily produce acoustic cavitation in a variety of liquids. It has been discovered that the damage to metal objects placed at the focus of an ESWL can be attributed to the action of liquid microjets produced by collapsing cavitation bubbles [A. J. Coleman *et al.*, *Ultrasound Med. Biol.* **13**, 69–76 (1987)]. Furthermore, Delius has shown that these shock waves can produce cavitation *in vivo* [Miller *et al.*, *J. Acoust. Soc. Am. Suppl.* **1** **83**, S 89 (1988)]. In this paper, a general review will be given of the mechanism of microjet formation in cavitation bubbles and arguments made that cavitation plays an important role in stone disintegration. A film will be shown of microjet formation in collapsing cavitation bubbles induced at low acoustic frequencies. [Work supported in part by NIH and ONR.]

### Contributed Paper

11:25

**GG6. A theoretical study of cavitation generated by an extracorporeal shock wave lithotripter.** Charles C. Church (Center for Biomedical Ultrasound, 319 Hopeman Hall, The University of Rochester, Rochester, NY 14627)

The intense acoustic wave generated at the focus of an extracorporeal shockwave lithotripter is modeled as the impulse response of a parallel RLC circuit. The shock wave consists of a zero rise time positive spike which falls to 0 at  $1 \mu\text{s}$  followed by a negative pressure component  $6 \mu\text{s}$  long with amplitudes scaled to  $+1000$  and  $-167$  bars, respectively. This

pressure wave drives the Gilmore–Akulichev formulation for bubble dynamics; the zero order effect of gas diffusion on bubble response is included. Results predict that a preexisting bubble in the micron range will expand to over 100 times its initial size,  $R_0$ , for  $240 \mu\text{s}$ , with a peak radius of  $1350R_0$ , then collapse very violently, emitting far UV or soft x-ray photons (black body assumption). Gas diffusion does not appreciably mitigate the violence of the primary collapse, but does significantly increase  $R_0$ , up to  $40 \mu\text{m}$ , as well as the duration of subsequent ringing, assuming no break up or production of microbubbles. The prediction of x-ray emission is of more academic than clinical concern because relatively few photons will be produced, and renal tissue exhibits a fairly low radio-sensitivity. [Work supported by NIH.]

**Session HH. Musical Acoustics III: Arthur H. Benade Memorial Session I: Horns, Strings, and Harmony**

Carleen M. Hutchins, Chairman

*Catgut Acoustical Society, 112 Essex Avenue, Montclair, New Jersey 07042*

Chairman's Introduction—8:30

*Invited Papers*

8:35

**HH1. Arthur H. Benade: The making of a musical acoustics researcher, theorist, and teacher.** Virginia Benade (3126 Woodbury Road, Cleveland, OH 44120)

The Rampur State Band, childhood consumption, and repairing World War II aircraft radios are only a few of the influences reported on here as having had some bearing on A. H. Benade's development as a musical acoustician.

9:05

**HH2. Arthur Benade—Synthesizer.** George Jameson (6214 38th Street, Kenosha, WI 53142)

Of the talents that led to Art Benade's considerable accomplishments in musical acoustics, one has escaped prominent mention—namely—his proclivity to synthesize. He had a strong ability to select ostensibly unrelated facts and thoughts from many disciplines, theoretical and practical, over a period of time. When these stored items began to fit together, Art was often able to discern a direction and reach the solution to an acoustical problem. The breadth of overall knowledge required to permit access to obscure relationships has often been itself obscured by the impact of Art's results. He called freely on branches of physics, chemistry, mathematics, and electronics not central to his work. Besides the sciences, he was well grounded in music, literature, and many phases of engineering. On a different level, he was a proficient machinist, mechanic, and tinkerer. Not least, he was a usable musical performer. He possessed a truly remarkable musical ear that was a major source of information. While many of the above abilities are requisite for any successful experimental physicist, Art extended his attentions further—to the user of musical instruments. Examples of the synthetic process will be cited—partly from the standpoint of the author, whose background is in manufacture and repair of instruments, and in routine professional performance and teaching of woodwinds.

9:35

**HH3. Fundamentals of piano scale design.** Earle L. Kent (2510 Riverview Place, Elkhart, IN 46516)

The determination of optimum piano string diameters, speaking lengths, and total lengths is complicated due to the interaction between the strings, their breaking strengths, available space for them, the energy they can accept from hammers, and the utilization of string energy to produce sound. Many economic, philosophical, and psychological factors in addition to the physical considerations are involved. No universally accepted definition of good piano tones has been established. Though it is not practical to lay down a set of specific rules for piano scale design, it is helpful to consider the relationships between involved parameters in order to guide the design, rebuilding, and repair of pianos. A set of string design relationships is given to serve as a benchmark in establishing piano string dimensions and as a basis for making comparisons in existing pianos. It seems reasonable to consider starting with well chosen string dimensions since the strings are the starting sources of piano tones.

10:05

**HH4. From twenty years of violin acoustics research.** Erik V. Jansson (Department of Speech Communication and Music Acoustics, Royal Institute of Technology (KTH), P.O. Box 700 14, S-100 44 Stockholm, Sweden)

A very fruitful year as research associate to Arthur Benade in 1967 started my professional work as a scientist. This year formed much of my way of thinking and future work on strings, some of the "highlights" of which are reviewed. Normal modes of plates of a violin under construction were mapped by means of hologram interferometry (with Molin and Sundin). Experiments with the air cavity, employing Fransson's Ionophone, disclosed a large number of air volume modes. Long-time-average spectra of quality judged violins implied that high level below 1 kHz, low at 1.3 kHz, and high around 2.5 kHz are favorable (with Sundberg and Gabrielsson). Normal modes of the complete violin were mapped with "speckle interferometry" and input admittance measurements, and implied that "equally strong" low-frequency modes and a marked "bridge hill" at 2.5 kHz

are positive for quality (with Alonso Moral). The influence of material properties and design of free plates were investigated in physical experiments and finite element calculations indicating that the thickness is the main factor (with Jakub Niewczyk, Molin, and Lindgren). The properties of plates in a simplified violin body are presently investigated (with Benedykt Niewczyk). I believe that the learnings from my year with Arthur Benade, thus, have further been spread during the last 20 years to my fellow colleagues in Sweden and guest researchers from Spain, Poland, and Australia.

10:35

**HH5. Listening to sound in rooms.** W. M. Hartmann (Physics Department, Michigan State University, East Lansing, MI 48824)

Professor A. H. Benade frequently remarked on the paradox involved in listening to musical sounds in rooms. He noted that the transmission path from the sound source to the listener is unreliable; its impulse response and its corresponding frequency response, are irregular, unpredictable, and unstable. Nevertheless, the listener obtains reliable information about the location of the source, about the tone color, and about the transient character. The paradox may be resolved in principle by postulating that the listener's nervous system performs certain temporal summation and differencing operations as well as binaural summation and differencing operations on the internal representation of the sound field. The present talk will review what is known from psychoacoustical experiments concerning these parallel summation and differencing operations with the goal of distinguishing between plausible and implausible models. Most striking has been recent progress on binaural processing where psychoacoustics and electrophysiology appear to be converging on a single model, based upon crosscorrelation in tuned channels. Of particular importance is the precedence effect, which, after 100 years of study, still creates surprises but still wants an adequate definition. [Work supported by the National Institutes of Health.]

11:05

**HH6. Art Benade at the Institut de Recherche et Coordination Acoustique/Musique in Paris.** René Causse (IRCAM, 31, rue Saint-Merri, 75004 Paris, France)

This presentation discusses Art Benade's influence on different instrumental acoustics projects at IRCAM, a musical research institution headed by composer Pierre Boulez. Art Benade was a member of IRCAM's original scientific advisory committee. New performance techniques, such as flute and trombone multiphonics, input impedance measurements of wind instruments and mutes for brasses were among the projects in which Art Benade made contributions, as a theoretical physicist and an adept experimenter, but also as an experienced musical performer and instrument maker. These multifaceted talents enabled him to communicate effectively with composers and performers, to understand musical problems in physical terms (and vice-versa), and to participate in every step of these projects. We will present the most recent results of some of these projects and discuss more particularly a new model of the "wah-wah" mute for the french horn.

11:35

**HH7. Reminiscences of Art Benade.** Harry B. Miller (Code 2131, Naval Underwater Systems Center, New London, CT 06320)

In the early 1950's, Art Benade gave a two-semester evening course on atomic physics. I was a member of the class. We soon got into after-class discussions of applied acoustics and these never really stopped until his death. Some of this will be recounted.

THURSDAY MORNING, 19 MAY 1988

GRAND BALLROOM B, 8:30 TO 11:15 A.M.

## Session II. Psychological Acoustics V: Psychoelectronics and Psychoacoustics in Impaired Versus Normal Listeners

David A. Nelson, Chairman

Hearing Research Laboratory, University of Minnesota, 2630 University Avenue, S.E., Minneapolis, Minnesota 55414

### Contributed Papers

8:30

**III. Gap discrimination in normal listeners and cochlear implant patients.** Robert V. Shannon and Donna L. Neff (Boys Town National Institute, 555 N. 30th Street, Omaha, NE 68131)

Gap discrimination was measured as a function of standard-gap duration and marker level in both normal-hearing listeners and in patients with Nucleus and Symbion cochlear implants. For normal listeners, the standard gap ranged from 0 to 400 ms and the broadband-noise markers were 30 to 90 dB SPL total power. For the implant patients, the range of

standard gaps and marker levels was similar, but the markers were either sinusoids or pulse trains. Normal listeners were tested under both monaural and dichotic (one marker to each ear) conditions. When expressed as  $\Delta T/T$ , the gap discrimination functions for both normals and implant patients are nonmonotonic with a peak at standard durations of 10–20 ms. Performance at the shortest standard durations is markedly poorer with dichotic presentation. The nonmonotonicity appears to reflect a change from a task limited by peripheral processes to one limited by more central processes. Cochlear implant patients exhibit a similar change. [Work supported by NIH.]

8:45

**II2. Aging effects on gap detection thresholds among normal and hearing impaired subjects.** Stanley A. Gelfand (Department of Communication Arts and Sciences, Queens College, Flushing, NY 11367), Jody Porrazzo (City University of New York, New York, 10036), and Shlomo Silman (Brooklyn College, Brooklyn, NY 11210)

Age differences between normal and hearing impaired groups have clouded the effects of hearing loss on temporal selectivity in many studies. This study attempted to address this problem by comparing gap detection thresholds (GDTs) between a normal hearing group and a group with primarily high-frequency sensorineural hearing losses, each containing subjects with similar age ranges. Transformed up-down estimates of 70.7% GDTs were used in a two-alternative forced-choice paradigm with feedback. Each run involved simultaneous estimates of two gap thresholds, using interleaved up-down procedures with random switching between them. Gap detection thresholds varied from approximately 3 to 16 ms among subjects in both groups. Across all subjects, GDTs were significantly correlated with age and with hearing sensitivity at 12 000 Hz but not with auditory thresholds at any other frequencies. Regression analyses revealed that age was the only significant predictor of GDTs among these subjects. The findings suggest that temporal processing is affected by age-related auditory changes that are not necessarily reflected as shifts of absolute sensitivity.

9:00

**II3. Difference limens (DLs) for phase in normally hearing subjects and subjects with unilateral cochlear hearing loss.** Brian C. J. Moore and Brian R. Glasberg (Department of Experimental Psychology, University of Cambridge, Downing Street, Cambridge CB2 3EB, England)

Complex tones containing the first 20 harmonics of 50, 100, or 200 Hz, all at equal amplitude, were used as stimuli. All of the harmonics except

one were added in cosine phase. The remaining harmonic started in cosine phase, but was shifted in phase halfway through either the first or the second of the two stimuli used in a 2-AFC trial. The task of the subject was to identify the stimulus containing the phase-shifted component. An adaptive procedure was used to measure DLs for phase for three normally hearing subjects and four subjects with unilateral cochlear impairments. When the normal ears were tested at a level of 70 dB SPL per component, DLs were smallest ( $2^{\circ}$ – $4^{\circ}$ ) for harmonics above the eighth and for the lowest fundamental frequency ( $F_0$ ). Changes in phase could not be detected for harmonic numbers below four at the lowest  $F_0$  and below 8–13 at the highest  $F_0$ . The DLs increased markedly with decreasing level, except for the highest harmonic, where only a small effect of level was observed. The DLs for the impaired ears were generally larger than those for the normal ears when the comparison was made at equal SPL but the discrepancy was reduced when the comparison was made at equal SL. For low harmonic numbers, a few cases were found where DLs were smaller for the impaired than for the normal ear of a given subject, even at equal SPL. This effect may be explicable in terms of the broader auditory filters of the impaired ears. The results will be compared to measures of temporal resolution and pitch discrimination using the same subjects. [Work supported by the Medical Research Council, U.K.]

9:15

**II4. Intensity discrimination at three frequencies by means of pulsed- and continuous-tone paradigms.** C. W. Turner, J. J. Zwislocki, and P. R. Filion (Communication Sciences and Disorders and Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-2280)

Difference limens (DL's) for the intensity of pure tones were measured with 2AFC pulsed-tone and continuous-tone paradigms at 500, 2000, and 6000 Hz across a broad range of sensation levels (SL's). In the pulsed-tone experiments, the pedestal was gated simultaneously with the increments, whereas, in the continuous-tone experiments the pedestal began 400 ms before the first observation interval, and continued throughout the trial until 400 ms after the termination of the second observation interval. The DL's obtained on five normal subjects were lower for the continuous-paradigm than for the pulsed-tone conditions for all test frequencies and SL's. The difference agrees with physiological results showing that the single-unit neural response to an increment is independent of whether the increment is presented to fibers that are adapted (continuous-tone experiment) or unadapted (pulsed-tone) experiment [Smith and Zwislocki, *Biol. Cyber.* (1975)] and also with the argument that the pulsed-tone paradigm involves the additional requirement of a "memory load" for subjects. However, according to our preliminary findings, the difference does not hold for all hearing-impaired subjects. [Work supported by NIH Grant NS24255.]

Break

9:30–9:45

9:45

**II5. Frequency resolution at equivalent sound-pressure levels in normal-hearing and hearing-impaired listeners.** David A. Nelson (Hearing Research Laboratory, University of Minnesota, 2630 University Avenue S.E. Minneapolis, MN 55414)

Forward-masked psychophysical tuning curves (PTCs) were obtained for multiple-level 1000-Hz probe tones from 15 normal-hearing and 23 hearing-impaired ears with cochlear hearing loss. Comparisons between normal-hearing and hearing-impaired PTC characteristics were made at equivalent masker levels near the tips of PTCs. The low-frequency slopes of PTCs from hearing-impaired listeners were not different from

those of normal-hearing listeners, i.e., hearing-impaired listeners did not demonstrate abnormal upward spread of masking when equivalent masker levels were compared. Only 8 of the 23 hearing-impaired listeners demonstrated abnormally broad PTCs, due exclusively to abnormally gradual high-frequency slopes to their PTCs. This demonstration of abnormal downward spread of masking was only observed in listeners with hearing losses greater than 40 dB HL. From these results, it would appear that some, but not all, cochlear hearing losses greater than 40 dB HL influence the sharp tuning capabilities usually associated with outer hair-cell function. The restriction of abnormal findings in this study to the high-frequency sides of PTCs alone is likely due to the use of equivalent masker-level comparisons, and also to the use of flat-envelope nonsimultaneous maskers rather than fluctuating-envelope simultaneous masker. [Work supported by NINCDS.]



10:00

**II6. Forward-masking properties of multicomponent signals in normal and hearing-impaired subjects.** Munah Kim, Christopher W. Turner, and Evan M. Relkin (Communication Sciences and Disorders and Institute for Sensory Research, Syracuse University, 805 S. Crouse Avenue, Syracuse NY 13244-2280)

The forward-masking properties of inharmonic complex stimuli were measured both for normal and hearing-impaired subjects. The signal was a single sinusoid of 1000 Hz. Signal threshold was obtained for six different maskers that varied in the number of components. Masking stimuli consisted of 1, 3, 5, 7, 9, or 11 components, logarithmically spaced in frequency surrounding the signal, and were presented at a fixed level of 80 dB SPL per component. An adaptive two-alternative forced choice method was used to determine thresholds. In normal-hearing subjects, the threshold for the signal decreased as the number of components was increased, demonstrating that signals with more components are less effective maskers. Results from hearing-impaired subjects indicated no decrease in threshold with increasing masking components, instead, the thresholds increased as more components were added to the first masker. [Work supported by the Deafness Research Foundation and the Whitaker Foundation.]

10:15

**II7. Growth of masking in listeners with cochlear impairments and impairments simulated by masking.** Søren Buus (Communication and Digital Signal Processing Center, 409 DA, Northeastern University, Boston, MA 02115) and Mary Florentine (Communication Research Laboratory, 133 FR, Northeastern University, Boston, MA 02115)

The higher-than-normal masked thresholds observed in many impaired listeners may be explained by nonlinear additivity of masking by the external masker and an hypothetical internal masker [Jesteadt *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S77 (1987); Humes *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S77 (1987)]. To test this hypothesis, Békésy tracking was used to measure growth-of-masking functions for 250-ms tones with frequencies at and above those of continuous low-pass or octave-band noise maskers. Results for four impaired listeners show that the minimum-effective-masker levels (MEMLs, above which masking occurs) are higher than normal for small masker-signal separations,  $\Delta f$ 's, but approach normal values for larger  $\Delta f$ 's. Above the MEMLs, masking grows 1 dB/dB or less. Results for simulated impairments are similar to those for real impairments, but MEMLs tend to be higher and masking often is less in simulated than in real impairments. These results indicate that nonlinear additivity of internal and external masking accounts for some, but not all, of the excessive spread of masking observed in many impaired listeners. [Work supported by NIH.]

10:30

**II8. Auditory filter width and consonant recognition for hearing-impaired listeners.** Judy R. Dubno and Donald D. Dirks (Division of Head and Neck Surgery, UCLA School of Medicine, Los Angeles, CA 90024-1794)

Abnormal frequency resolution is often assumed to contribute to reduced speech recognition by hearing-impaired listeners. However, a strong association between these two conditions secondary to hearing impairment has not been established, perhaps because both are degraded by threshold elevation. Thus, in this experiment, which examined auditory filter effects on consonant recognition, presentation levels of speech

stimuli for each subject were selected on the basis of predictions from Articulation Index theory, in order to control for threshold sensitivity differences among the normal-hearing and hearing-impaired listeners. Estimates of filter width for 25 hearing-impaired subjects (calculated from masked thresholds measured in broadband noise as function of the width of a spectral notch) revealed auditory filter widths that increased with the threshold for the probe. Consonant place perception for most hearing-impaired subjects was nearly equivalent to results for normal-hearing listeners when threshold differences were controlled. These results suggest that frequency resolution and speech recognition may coexist as independent consequences of threshold elevation. [Work supported by grants from NINCDS and the Deafness Research Foundation.]

10:45

**II9. Auditory temporal summation in normal-hearing and hearing-impaired cat.** Janet M. Solecki and George M. Gerken (Callier Center for Communication Disorders, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)

Auditory temporal summation functions were measured for five cats before and after they were exposed to a 2-kHz tone at 110 dBA for 48 h. Ten digitally generated stimuli with overall durations ranging from 8.32 to 275 ms were used. Stimulus frequency was 6.25 kHz. Six stimuli consisted of single tone bursts and the remaining four contained multiple tone bursts. Twelve acceptable thresholds for each stimulus were obtained from each animal before and after exposure. Threshold measurements began again no sooner than 27 days postexposure. Pre- and postexposure audiograms were also obtained and mean permanent threshold shift at 6.25 kHz was 32.6 dB. Pre- and postexposure temporal summation functions were characterized by the slope of the regression lines for threshold versus log duration. The mean slope in dB per decade of duration was  $-6.6$  preexposure and  $-3.8$  postexposure. The slope for normal-hearing cat is very similar to the slope of  $-6.1$  that was obtained in a companion study that used normal-hearing human subjects.

11:00

**II10. Some parameters of automatic audiometric thresholds.** I. M. Young and L. D. Lowry (Department of Otolaryngology, Jefferson Medical College of Thomas Jefferson University, Philadelphia, PA 19107)

Threshold measurements were made by two automatic audiometers; (1) continuously variable frequency tracing audiometer with an attenuation step of 0.25 dB; (2) continuously changing discrete individual frequency tracing audiometer with an attenuation step of 1.0 dB. Five trained subjects with bilateral normal hearing, four subjects with unilateral cochlear lesion, and four subjects with unilateral retrocochlear lesion were tested. For both pulsed and continuous tones, there was no observable difference in thresholds between two audiometric measurements for subjects with bilateral normal hearing and for ears with normal hearing in subjects with unilateral hearing impairment. The ears with cochlear lesion indicating abnormal rapid adaptation showed slightly greater separation between pulsed and continuous tone thresholds by continuously variable frequency tracings than by continuously changing discrete frequency tracings. For the ears with retrocochlear lesion indicating abnormal slow type of adaptation, much greater separation was demonstrated by the former. These findings were discussed with and related to parameters affecting thresholds; slow attenuation step in size, threshold at equilibrium (stabilized threshold), and lateral frequency spread of adaptation.

## Session JJ. Structural Acoustics and Vibration III: Plates and Shells and Radiation Problems

Sabih I. Hayek, Chairman

*Engineering Science and Mathematics, Applied Research Laboratory, The Pennsylvania State University, University Park, Pennsylvania 16802*

## Contributed Papers

8:30

**JJ1. An analytic model for vibration of turbomachinery rotor blades.** James P. Charpic and Courtney B. Burroughs (Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, PA 16804)

An analytic model for the vibration response of turbomachinery rotor blades that includes the effects of variations in blade thickness, curvature of the blade edges, and variations in the curvature of the blade middle surface in two dimensions is developed. Mathematical fits to both surfaces of the blade are generated in terms of polynomial equations with constant coefficients. The polynomial equations for the blade surfaces are then used in shell equations, based on Love's thin shell assumptions, to generate differential equations governing the vibration motion of the blade. The coefficients in the blade-fitting polynomials act as parameters in the equations of motion for the blade, so that equations for a new blade can be developed easily by changing the coefficients in the governing equations to those generated for the new blade. An approximate method of solution for the resonant response of a sample blade is presented.

8:45

**JJ2. Modification of Lamb wave spectrum in fluid coupled plate.** S. I. Rokhlin (Department of Welding Engineering, The Ohio State University, 190 West 19 Avenue, Columbus, OH 43210), D. E. Chimenti (Materials Laboratory, Wright-Patterson Air Force Base, Dayton, OH 45433), and A. H. Nayfeh (Department of Aerospace Eng., M.L. 70, University of Cincinnati, Cincinnati, OH 45221)

It has been accepted previously that for most practical cases when a plate is immersed in a liquid the Lamb wave spectrum is only slightly perturbed due to energy loss related to leakage of the wave from the plate to the liquid. In contrast it was recently discovered by D. E. Chimenti and A. H. Nayfeh [Appl. Phys. Lett. 49 (9) (1986)] that for leaky Lamb waves in a graphite epoxy composite plate in water a strong anomaly appears in the topology of the  $S_0$  mode as a function of frequency. Although it was caused by the fluid, the reason for this anomaly was still not clear. The nature of the anomalous behavior of leaky Lamb waves is discussed in this paper. It is shown that for an understanding of the spectrum topology, the full complex spectrum of propagating and nonpropagating modes should be considered. It is demonstrated that the topological changes in the  $S_0$  mode are related to its nearness to the first complex branch of the spectrum. Increasing liquid density leads to interaction between these modes and mutual interchange between their branches in the complex plane. It is shown that, in general, as the density of the immersion liquid increases from zero to infinity the complex spectrum of plate modes gradually transforms from a spectrum of Lamb waves in a free plate to a spectrum of waves in a plate clamped on its surfaces with slip (boundary conditions  $\sigma_{xy} = 0$ ,  $U_y = 0$ ).

9:00

**JJ3. The solution of fluid-structure interaction problems using quadratic, doubly curved finite elements.** Joseph J. Shirron and Luise Schuetz (Naval Research Laboratory, Code 5134, 4555 Overlook Avenue, Washington, DC 20375-5000)

A model to solve coupled fluid-structure interaction problems, such as acoustic scattering from an elastic body, is described. This model uses doubly curved, nine-noded, isoparametric finite elements over which the pressure is taken to vary quadratically. Results from this model are compared with another computer model (NASHUA) that is based on planar finite elements over which the acoustic pressure is assumed to remain constant. Several examples are presented to demonstrate the reduction in the number of elements needed for accurate modeling. Comparison of solution time and accuracy is also given.

9:15

**JJ4. Acoustical power radiation from cylindrical shells, with or without stiffeners: Understanding of phenomena and practical aspects.** B. Laulagnet, J. L. Guyader, and C. Lesueur (Laboratoire Vibrations-Acoustique, Institut National des Sciences Appliquées de Lyon, 69621 Villeurbanne Cedex, France)

An analytical study of radiation by finite, stiffened cylindrical shells is presented. The model of stiffener utilized allows one to treat hollow cross-section cases encountered in industry. The point driving force is on the stiffener or directly on the shell and can be radial, tangential, or longitudinal. Fluid and structure equations are solved with the modal analysis, using the *in vacuo* nonstiffened basis. This leads to the calculation of radiation impedances and mass and stiffness generalized terms of stiffeners. Theoretical results are presented both in air and in water, in radiated power, radial quadratic velocity, and radiation factor. Mechanical modal cross-coupling influence is shown as well as the stiffening influence on the in water vibroacoustic shell behavior. Parameters like rigidity, number and distribution of stiffeners, shell and stiffener structural damping, location, and force type are studied in air. Finally means for noise reducing in air are deduced. [Work supported by the Ministère de la Défense, Contrat D.R.E.T. No. 85065.]

9:30

**JJ5. Sound radiation from a vibrating body in motion.** Xiao-Feng Wu and Adnan Akay (Department of Mechanical Engineering, Wayne State University, Detroit, MI 48202)

Past studies of acoustic radiation problems have been concerned with sound waves generated by either a vibrating body or a moving point source with no dimensions of its own. This paper presents a method for determining the acoustic field radiated from an arbitrary object undergoing, simultaneously, a harmonic oscillation and a spatial motion with respect to the surrounding fluid medium. At this stage of the investigation, the turbulence and flow distortion caused by the motion of the vibrating body are neglected. The method is based on the assumption that the surface of the vibrating body can be described in terms of a distribution of simple monopole and dipole sources. Utilizing a linear coordinate transformation, one is then able to reduce the problem to that of radiation from a stationary source for which the analytical solution has been obtained [Morse and Ingard, *Theoretical Acoustics* (Princeton U.P., Princeton, NJ, 1986)]. The total acoustic field is the sum of contributions from all the simple sources on the surface of the moving object. The method is illustrated using a vibrating sphere moving with either constant or time-

dependent velocity profiles relative to the surrounding medium. Extension of this method to impulsive motions is discussed. [Work supported by NSF and WSU Institute for Manufacturing Research.]

9:45

**JJ6. Leaky Lamb wave inspection of thin plates.** Peter B. Nagy and Laszlo Adler (Department of Welding Engineering, The Ohio State University, Columbus, OH 43210)

A single-transducer immersion technique is introduced to study dispersion of leaky Lamb waves in different plates, such as fiber-reinforced composite laminates and adhesive lap joints. The plate under study is interrogated by a broadband pulse from both sides when placed between the transmitting-receiving transducer and a perpendicularly aligned mirror. The transmitted peaks in the spectrum of the received echo from this mirror correspond to double mode-converted leaky Lamb wave resonances. These peaks are measured at different angles of incidence, i.e., at given phase velocities to get velocity dispersion curves. Experimental results are presented to demonstrate the accuracy and versatility of this very simple measuring technique.

10:00

**JJ7. Transient structural vibration created by a laser beam.** Yousef A. Abo-Mossallam (Faculty of Engineering, El-Mansoura University, Egypt)

Experimental work is conducted to investigate the transient vibration of a structure by an argon laser beam. The conversion of laser light into mechanical energy has been studied through focusing laser beam to vaporize material from a target attached to the structure. The rapid ejection

of target material results in an impulsive reaction transmitted to the structure. Different target materials are tested to enhance the efficiency of energy conversion. A clamped beam and a cantilever beam are used in the experiments and the measured natural frequencies are compared favorably with the calculated ones. Beam splitters and multiple targets are used to test the concept of multipoint simultaneous impulsive loading.

10:15

**JJ8. Sound radiation control for two plates assembled in an L shape.** Pascal F. Millot (Laboratoire des Sciences de l'Habitat, Ecole Nationale des Travaux Publics de l'Etat, Rue Maurice Audin, 69 120 Vaulx-en-Velin, France) and Jean-Louis Guyader (Laboratoire Vibrations Acoustique, Institut National des Sciences Appliquées, 69 621 Villeurbanne Cedex, France)

Reduction of sound radiated in a cavity by an L-shape structure excited by mechanical forces, can—in some instances—be achieved by taking advantage of the phenomena that are unique to that geometrical shape. Instead of relying only on the control of the velocity of the structure to reduce the sound radiated, it will be shown that it is possible to select modes of the assembled plates that have a low-radiation impedance, by positioning correctly the mechanical excitation forces. At low frequencies, improvements can reach several decades for the mean-squared pressure inside the cavity for identical plates, while above the limit frequency of the cavity no improvement should be expected with this method. Below the limit frequency, improvements will greatly depend on the respective dimensions of the plates, as well as on the values for the first resonant frequency of the plates and of the cavity. Theoretical results, derived from a modal approach for the cavity and the plates, as well as experimental results obtained for plaster plates in laboratory, will be presented.

THURSDAY MORNING, 19 MAY 1988

WEST BALLROOM B, 8:30 A.M. TO 12:05 P.M.

### Session KK. Underwater Acoustics V: Bottom Interaction

Barry J. Uscinski, Chairman

*Department of Applied Mathematics and Theoretical Physics, Cambridge University, Cambridge CB3 9EW, United Kingdom*

Chairman's Introduction—8:30

#### *Contributed Papers*

8:35

**KK1. Computation of the Biot drag and virtual mass coefficients for a three-dimensional pore space.** B. Yavari and A. Bedford (Department of Aerospace Engineering and Engineering Mechanics, The University of Texas, Austin, TX 78712)

Application of the Biot theory to the study of the acoustic waves in ocean sediments requires evaluation of the drag and virtual mass coefficients. These coefficients depend on the frequency and microstructure of the sediments. Recently, a method was developed to evaluate these coefficients by Bedford *et al.* [J. Acoust. Soc. Am. 76, 1804–1809 (1984)]. The method requires solving for the motion of the fluid in the pores when the pore walls are subjected to an oscillatory motion. The finite element method has been used to determine the fluid motion. The drag and virtual mass coefficients have been determined for a microstructure composed of spherical grains saturated with water. The qualitative behavior of the coefficients is shown to be similar to the two-dimensional results present-

ed previously [J. Acoust. Soc. Am. Suppl. 1 82, S121 (1987)]. The behavior is also shown to be qualitatively similar to that of an exact solution for the case of straight cylinder pores obtained by Biot and other investigators. Wave velocity and attenuation of compressional waves calculated based on these coefficients are shown to agree reasonably well with experimental data. [Work supported by ONR.]

8:50

**KK2. Shear wave velocity in surficial marine sediments: A comparison of *in situ* and laboratory measurements.** Michael D. Richardson, Enrico Muzi, and Luigi Troiano (SACLANT Underwater Research Centre, I-19026 La Spezia, Italy)

*In situ* and laboratory measurements of shear wave velocity were compared for a wide variety of shallow-water marine sediment types found in

the vicinity of La Spezia, Italy. Both types of measurements utilized pulse techniques, with shear waves generated and received by bimorph ceramic bender elements. Measurements were made between 200 and 1000 Hz. *In situ* measurements and sediment collections were made by scuba divers to minimize sediment disturbance. Values of shear wave velocity were low ( $< 100$  m/s) for all sediment types sampled. Typical values ranged from 20 m/s in high porosity clays to 50 m/s in sandy sediments. Shear wave velocities measured by *in situ* and laboratory techniques were not significantly different, suggesting that shear wave velocities measured on carefully collected core samples of surficial sediments need not be corrected to *in situ* conditions. Apparently, changes in shear modulus, which result from sample disturbance during coring, and changes in pressure and temperature after collection are small compared to the natural variability of surficial sediment shear wave velocity.

9:05

**KK3. Preliminary results from a shallow water bottom interaction experiment in the Gulf of Mexico.** James F. Lynch, George V. Frisk, and Subramaniam D. Rajan (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

In order to help assess our present capability to predict the details of sound propagation in shallow water environments, a cw acoustics experiment was performed at a site in the Gulf of Mexico near Corpus Christi, TX. This site was chosen because it was both relatively simple acoustically and previously well-studied. Indeed, the site chosen was the same as was studied by Rubano in an earlier experiment [L. A. Rubano, *J. Acoust. Soc. Am.* **67**, 1608–1613 (1980)]; moreover, a compilation of archival records (seismic data, etc.) by a NORDA group gave a second geoacoustic model to complement Rubano's [J. E. Matthews, P. J. Bucca, and W. H. Geddes, NORDA Report 120, June 1985]. The data were taken at 50 and 140 Hz on radial tracks typically 5 km long, and sampled at least at the spatial Nyquist frequency (every half-wavelength in range), so that each data track was effectively a 5-km synthetic aperture array. The data were then Hankel transformed to display the normal mode structure of the waveguide [G. V. Frisk and J. F. Lynch, *J. Acoust. Soc. Am.* **76**, 205–216 (1984)], and the mode structure was in turn inverted to yield a geoacoustic model [S. D. Rajan, J. F. Lynch, and G. V. Frisk, *J. Acoust. Soc. Am.* **82**, 998–1017 (1987)], which we compare to the NORDA and Rubano models. Comparisons of pressure field predictions, modal structure predictions, and geoacoustic models are discussed. [Work supported by ONR.]

9:20

**KK4. Propagation loss modeling of the seabed using both shear and permeability effects.** Mohsen Badiy and Tokuo Yamamoto (Geoacoustics Laboratory, University of Miami, Miami, FL 33149)

A complete normal mode propagation model is used to predict transmission loss coefficients for two previously reported data sets, one from the Nova Scotian Shelf and one from the Mediterranean Sea. This model employs both permeability and shear modulus of the seabed through formulations of Biot–Willis theory of acoustic waves in poro-elastic, anisotropic media. The sediments were modeled using the reported geological and core data. It was found that both shear and hydraulic (permeability) properties of the sediments contribute to the propagation loss. At low frequencies waves penetrate deep into the sediments and are affected by the higher rigidity of the lower sediment layers. Higher frequencies penetrate only into surficial sediments where the permeability is the dominant influence in the attenuation mechanism. [Work supported by ONR.]

9:35

**KK5. Low-frequency scattering from the upper crust.** Orest Diachok, John Shaffer (Code 5120, Naval Research Laboratory, Washington, DC 20375), and Laurel Henderson (SACLANTCEN, LaSpezia, Italy)

The rms roughness of sediment-free basalt at a site east of the East

Pacific Rise [measured by Lonsdale *et al.* (1980)] was computed versus averaging length for predictions of scattering loss versus Fresnel zone dimensions. The source–receiver geometry replicated previously, reported measurements of the reflection coefficient versus grazing angle at a nearby, sediment-free site [Diachok *et al.* (1986)]. The computed roughness increased slowly with decreasing grazing angle ( $\theta$ ) at large  $\theta$ , being about 8 m (at 20 Hz) at  $60^\circ$ , and increased rapidly with decreasing grazing angle for small  $\theta$ , being about 38 m (at 20 Hz) at  $10^\circ$  along a track that paralleled the ridge axis. The effective rms roughness inferred from the reflectivity data for  $50^\circ < \theta < 70^\circ$  was 6 m (at 20 Hz along a parallel track), approximately consistent with the inference from the deep tow bathymetric data. At large  $\theta$ , the effective roughness  $r$  is much smaller than the wavelength  $\lambda$ , permitting use of Eckart scattering theory. At small  $\theta$ , however,  $r$  and  $\lambda$  are comparable, violating the Kirchhoff approximation. Implications of these observations and other calculations for modeling reflectivity from sediment-free/thinly sedimented basalt, a prevalent feature of the Pacific, will be discussed.

9:50

**KK6. Selected examples of Scholte wave propagation in deep and shallow water.** Craig A. Fisher and Hassan B. Ali (Naval Ocean Research and Development Activity, NSTL, MS 39529-5004)

The Naval Ocean Research and Development Activity recently completed a series of three experiments to measure very low-frequency (VLF) seismo-acoustic propagation in both shallow and deep water marine environments. A vertical hydrophone array and ocean bottom instruments were deployed in all three experiments. The ocean bottom instruments contained both hydrophone and three-component geophone sensors. In the single shallow water experiment, measurements were also made with buried three-component geophones. We compare the VLF propagation characteristics in deep and shallow water seen with ocean bottom and water column sensors. The importance of Scholte wave propagation in accounting for both signal and ambient noise energy at near bottom receivers is shown for selected frequencies using both measured and modeled results. Necessary environmental conditions for Scholte wave excitation are also examined.

10:05

**KK7. Gaussian beam method for underwater acoustic reflection from an elastic bottom.** I. T. Lu, X. J. Gao, and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, NY 11735)

The Gaussian beam method furnishes a potentially useful numerical algorithm for modeling propagation in an inhomogeneous ocean environment. However, the paraxially approximated beams in the heuristically discretized stack suffer from spectral deficiencies that are problem dependent and make *a priori* predictability uncertain. Understanding these deficiencies may provide clues as to how to deal with them. The present test environment is a line source excited homogeneous ocean with a solid bottom that gives rise to two distinct critical reflections associated with the compressional ( $P$ ) and vertical ( $SV$ ) waves, respectively. Stacks of narrow beams, which have localizing advantages, have previously been found incapable of accounting for the critical angle (head wave) effects when the relevant wave phenomena involve only  $P$  or  $SV$  (horizontal shear), respectively [I. T. Lu, L. B. Felsen, and Y. Z. Ruan, *Geophys. J. R. Astron. Soc.* **89**, 915–932 (1987)]. It is confirmed here that this situation persists in the presence of  $P$ – $SV$  coupling. As before, very wide paraxial beams or narrow beams with *full* spectra remove these effects, but tracking the latter is computationally more difficult, while tracking the former is inconvenient and uncertain for successive encounters with an environment. Various alternative options are examined to cope with these difficulties. [Work supported by NOSC.]

10:20

**KK8. The modeling of a shallow-water wedge using complex rays and steepest-descent integration.** Evan K. Westwood (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713)

A new method for computing propagation loss in a shallow-water wedge with a penetrable bottom will be presented. The method is based on the representation of the total field as a sum of ray fields. Each ray field is expressed as a contour integral over plane waves in the complex  $\theta$  plane, where  $\theta$  is incident angle. Saddle points (possibly complex) are found for each ray, and steepest descent path integration is required when the ray strikes the bottom interface near the critical angle. Lateral wave contributions are also included. This model has been applied to the upslope benchmark problems proposed at the 112th Meeting. The results are indistinguishable from those reported by Jensen at the 113th Meeting using Evans' two-way coupled mode model (generally considered to be the most accurate). The computation time for 400 points on a horizontal line is on the order of 5 min. Unlike coupled mode and parabolic equation models, the computation time using this ray model *decreases* as the frequency is raised. [Work supported by the ONR contract N00014-87-K-0346.]

10:35

**KK9. Transient Green's tensors for a layered solid half-space with different interface conditions.** Shuchu Ren, Nelson N. Hsu, and Donald G. Eitzen (National Bureau of Standards, Gaithersburg, MD 20899)

To understand the influence of the interface conditions on the behavior of transient waves in a structure, the Green's tensor for an isotropic layer overlay on an isotropic half-space is solved using the "generalized ray" expansion technique and Willis inversion method. This inversion method has many advantages over the Cagniard-de Hoop inversion method. First, the displacements rather than the potentials are used and the derivations are greatly simplified; second, self-similar sources are assumed and a detailed discussion of the structure of a complicated algebraic function is avoided; and third, the derivatives of the Green's tensor about spacial coordinates are easily obtained. A FORTRAN code for computation of the Green's tensors was developed. Both the cases of a welded and a liquid coupled interface are considered. Computed results show that changes of the interface condition have a great influence on the behavior of transient waves. For example, when the interface conditions vary from welded to liquid coupled, some of the first few head waves nearly vanish while the first few regular reflected rays keep unchanged. This phenomenon can be used to infer the quality of the interface bond of materials in ultrasonic NDE.

10:50

**KK10. Propagation effects of a transition layer at the ocean/sediment interface.** Michael D. Collins, Stanley A. Chin-Bing (NORDA Numerical Modeling Division, NSTL, MS 39529), and Joseph E. Murphy (Department of Physics, University of New Orleans, New Orleans, LA 70148)

In this paper, the results of an investigation into the propagation effects of a transition layer in the uppermost portion of the ocean sediment are presented. The layer of sediment directly beneath the ocean bottom is partially saturated with water and is close to the temperature of the water. Due to these saturation and thermodynamic effects, the discontinuity in the speed of sound at the ocean/sediment interface is typically about a half of a percent. Since the deeper layers are insulated from the ocean and are not saturated with water, one would expect the sound speed  $c_0$  deep within the sediment to depend only on the depth  $z$  below the ocean surface to first order. Thus a reasonable formula for sediment sound speed is  $c_s(z, d) = [\alpha c_w(d) - c_0(d)]e^{(\alpha - z)/h} + c_0(z)$ , where  $\alpha \geq 1$ ,  $c_w$  is the ocean sound speed,  $d$  is the ocean depth, and  $h$  is the thickness of the transition layer. When attenuation is present in the transition layer, propagation loss increases with  $h$  since a thicker transition layer results in longer ray path lengths within the lossy region. [Work supported by ONR and NORDA.]

11:05

**KK11. The outgoing solution of the Klein-Gordon equation and the effects of dispersive sediments on pulse propagation in the ocean.** Michael D. Collins (NORDA Numerical Modeling Division, NSTL, MS 39529)

The parabolic equation method, which has been used as a frequency domain tool in underwater acoustics for several years, is also effective for use in the time domain [B. E. McDonald and W. A. Kuperman, *J. Acoust. Soc. Am.* **81**, 1406-1417 (1987)]. In this paper, some recent progress in the development of the outgoing solution of the Klein-Gordon equation is described and some calculations are presented. It was found that very small amounts of sediment dispersion result in significant distortion of even the water-borne portion of signals. This surprising finding should help resolve a controversy regarding the relationship between attenuation and frequency in sediments, which has long been accepted to be linear based on several experiments. In a recent paper [Zhou *et al.*, *J. Acoust. Soc. Am.* **82**, 2068-2074 (1987)], a different relationship is proposed based on the analysis of arrival times. Despite the fact that attenuation and dispersion are very intimately related, dispersion is neglected in the analysis. [Work supported by ONR and NORDA.]

11:20

**KK12. Bottom classification using backscattering at vertical incidence.** Darrell R. Jackson (Applied Physics Laboratory, College of Ocean and Fishery Sciences, University of Washington, Seattle, WA 98195) and Eric Nesbitt (Boeing Commercial Airplane, M/S 01-41, Seattle, WA 98124)

A technique has been developed for measurement of bottom geacoustic properties using backscattering at vertical incidence. The parameters of a scattering model are determined by a maximum-likelihood fit to received intensity as a function of time. The procedure provides measurements or bounds for the following sediment bulk acoustic properties; impedance, absorption coefficient, and volume scattering strength. In addition, a measure of interface roughness can be obtained if the coherent component of received intensity is negligible. The method has been tested using high-frequency data from shallow silty and clayey sites. [Work supported by ONT and NORDA.]

11:35

**KK13. Time spread of bottom interacting signals from an abyssal plain environment.** Paul J. Vidmar<sup>1</sup> (Science Applications International Corporation, 1710 Goodridge Drive, McLean, VA 22102)

Measured signals from explosive sources are examined to determine some characteristics of the time spread introduced by bottom interaction in an abyssal plain area. Broadband (10-1600 Hz) acoustic data are examined that sample the transition from sediment penetrating paths that refract within the sediment to those that reflect from the basalt substrate beneath the sediment. Different characteristic time spreads at high and low frequencies were observed. In a high-frequency band (1500-1600 Hz), the time spread is relatively short ( $\sim 200$  ms), while in a low-frequency band (50-150 Hz) the time spread is very long ( $> 0.5$  s). The short duration of the high-frequency time spread is consistent with reflection from layer interfaces in the sediment. The long duration of the time spread at low frequencies is consistent with interaction with the substrate (e.g., scattering). [Work supported by the NORDA Bottom Interaction Project and the ONR AEAS Program.] <sup>1</sup> On leave from Applied Research Laboratories, The University of Texas at Austin.

11:50

**KK14. The low-frequency reflection coefficient of a gassy sediment.** K. E. Gilbert and J. R. Zagar (The National Center for Physical Acoustics, University of Mississippi, University, MS 38677)

Narrow-band propagation measurements have been made in a shallow pond with a gas-saturated bottom [L. E. Murray and K. E. Gilbert, *J. Acoust. Soc. Am. Suppl.* 1 **80**, S63 (1986)]. A normal-mode analysis of the data was used to extract plane-wave reflection coefficients for the bottom. Between 600 Hz (the cutoff frequency for the first mode) and 2000 Hz, the reflection coefficient  $R$  varies approximately linearly with

frequency. The end points are  $R = -0.91 \pm 0.02$  at 600 Hz and  $R = -0.82 \pm 0.02$  at 2000 Hz. These values for  $R$  are similar to those that have been obtained in broadband experiments [e.g., Jones *et al.*, *J. Acoust. Soc. Am.* **36**, 154-157 (1964)]. Physical mechanisms that may control the low-frequency reflection coefficient of a gassy sediment will be discussed.

THURSDAY MORNING, 19 MAY 1988

418/420 ROOM, 9:30 A.M.

## Meeting of Accredited Standards Committee S12 on Noise

to be held jointly with the

### Technical Advisory Group for ISO/TC 43/SC 1 Noise

W. Melnick, Chairman S12

*Ohio State University, University Hospital Clinic, 456 Clinic Drive, Columbus, Ohio 43210*

H. E. von Gierke, Chairman, Technical Advisory Group for ISO/TC 43/SC 1

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

**Standards Committee S12 on Noise.** Working group chairs will report on their progress under the plan for the production of noise standards. The interaction with ISO/TC 43/SC 1 activities will be discussed.

THURSDAY MORNING, 19 MAY 1988

EAST BALLROOM A, 10:45 A.M. TO 12:15 P.M.

## Session LL. Structural Acoustics and Vibration IV: Piezoelectric and Damping Materials

Donald Ricketts, Chairman

*Raytheon Company, 1847 West Main Road, Portsmouth, Rhode Island 02871-1087*

### *Contributed Papers*

10:45

**LL1. Temperature dependence of the electromechanical properties of ceramic polymer composite materials for hydrophone applications.** Kurt M. Rittenmyer (Underwater Sound Reference Detachment, Naval Research Laboratory, P.O. Box 568337, Orlando, FL 32856-8337)

The real and imaginary parts of the complex elastic, dielectric, and piezoelectric properties of two types of commercial piezoelectric composite materials were studied as a function of temperature from  $-70^\circ\text{C}$  to  $+50^\circ\text{C}$  using an electrical resonance technique described previously [*J. Acoust. Soc. Am. Suppl.* 1 **82**, S101 (1987)]. These materials consist of ceramic particles embedded in a polymer material and possess improved hydrostatic sensitivity as well as mechanical flexibility in comparison with conventional piezoelectric ceramics. Measurements of the temperature dependence of the hydrostatic piezoelectric coefficients have also been performed. The interrelationship of the properties is complicated by the presence of the glass transition of the elastomeric phase of the composite material where sharp changes in the electromechanical transition causes

typical relaxation behavior in the complex elastic coefficients and a peak in the dielectric properties. The transition was studied further by the use of differential scanning calorimetry in order to corroborate the results of the resonance measurements. The results demonstrate some difficulties in using these particular materials and indicate that other polymeric materials should be investigated for use in these composite materials. [Work supported by ONR.]

11:00

**LL2. Lower symmetry of the elastic and piezoelectric tensors.** D. F. Nelson (Department of Physics, Worcester Polytechnic Institute, Worcester, MA 01609)

A long wavelength lattice dynamical calculation of the elastic and piezoelectric tensors that does *not* use the adiabatic approximation yields linear coupling to rotation as well as to strain in dynamic interactions.

This leads in the general anisotropic (triclinic) case to 45 independent elastic components (rather than 21 as believed for over a century) and to 27 independent piezoelectric components (rather than 18). The new contributions to these tensors result from the inertial effects of the optic modes and are proportional to the square of the frequency. Thus they vanish for static interactions and so restore the traditional symmetries in this limit. The new tensor components create an antisymmetric part to the stress tensor; it produces a torque that is balanced by a change in the internal angular momentum density of the optic modes. The new tensor components will be measurably large only for frequencies comparable to optic mode frequencies, which, in most crystals, are well beyond the acoustic region. However, near a second-order phase transition, the frequency of a soft optic mode approaches zero. In some cases, such as chloranil, the soft mode has been observed in the hypersonic region as a distinct (underdamped) resonance. In such cases, the new tensor components can be expected to be measurably large.

11:15

**LL3. Thermopiezoelectric equations of high-frequency vibrations of ceramic shells.** M. Cengiz Dökmeci (Istanbul Technical University, P. K. 9, Taksim, Istanbul 80191, Turkey)

In view of the author's recent review article [Shock Vib. Dig. 20(2), 3-20 (1988)], this study is carried out for the aim of a systematic and consistent derivation of the governing equations of vibrations of piezoceramic shells. First, a variational principle is derived so as to describe all the three-dimensional equations of thermopiezoelectricity. Then, the mechanical displacements, electric potential, and temperature increment of piezoceramic shell are expanded in power series of the thickness coordinate. Next, a system of two-dimensional equations of successively higher-orders approximation is derived for vibrations of thermopiezoelectric shell by use of the variational principle together with the series expansions. The system of equations governs all the types of coupled vibrations of piezoceramic shell [cf. R. D. Mindlin, *Int. J. Solids Struct.* 10, 625-637 (1974)]. Also, special motions and geometry of piezoceramic shell are indicated. Finally, a theorem of uniqueness is proved for solutions of the governing equations of thermopiezoelectric shell. [Work supported in part by the United States Army through its European Research Office.]

11:30

**LL4. Evaluation of the complex coefficient  $\hat{\xi}_{33}^D$  of thick film piezoelectric polymers.** Donald Ricketts and Thomas Howarth (Raytheon Company, 1847 West Main Road, Portsmouth, RI 02871-1087)

A model for evaluating  $\hat{\xi}_{33}^D$  is presented using an impact testing technique [J. Acoust. Soc. Am. Suppl. 1 67, S32 (1980)]. In the form of a consolidated bare ring stack, the piezoelectric polymer (PVF<sub>2</sub>) test specimen is sandwiched between two identical loading masses. A threaded tie rod passes through the center of the entire assembly and is fastened at each end by means of a nut. This arrangement facilitates evaluating  $\hat{\xi}_{33}^D$  as a function of frequency and compressive stress. By impact testing this composite longitudinal vibrator, its resonance frequency ( $f_R$ ) and mechanical storage factor ( $Q_M$ ) may be measured. From these measured quantities and the mathematical model of the test resonator, the complex elastic compliance coefficient  $\hat{\xi}_{33}^D$  may be evaluated. The expressions for evaluat-

ing the real and imaginary parts of  $\hat{\xi}_{33}^D$  are given. Experimental results obtained at room temperature are presented for  $\hat{\xi}_{33}^D$  versus compressive prestress at two nominal audio frequencies.

11:45

**LL5. Development of polyurethane encapsulants with improved resistance to seawater exposure.** G. M. Stack, J. M. Miller (Underwater Sound Reference Detachment, Naval Research Laboratory, P. O. Box 568337, Orlando, FL 32856-8337), and E. Y. Chang (American Cyanamid Co., Bridgewater, NJ 08807)

Polyurethanes have been widely used as underwater acoustic encapsulants for over 20 years. In an attempt to produce polyurethane encapsulants with improved resistance to seawater exposure, polyurethane systems containing aliphatic isocyanate groups were investigated. Several commercial prepolymers containing aliphatic isocyanate groups were obtained for evaluation. Initially, low-temperature cure systems were developed to cure these prepolymers. It was found that the physical and acoustic properties of the resultant polyurethanes depended significantly on the composition of the cure formulation. The effect of prepolymer composition on the properties of the cured urethanes was then investigated. In particular, the molecular weight and isocyanate content of the prepolymers were varied. Several urethane systems suitable as underwater encapsulants were developed. These urethanes were subjected to long-term aging in seawater. It was found that this exposure to seawater did not significantly affect the dynamic mechanical properties of these polyurethanes. [Work supported by Sonar Transducer Reliability Improvement Program (STRIP).]

12:00

**LL6. Influence of fillers on constrained-layer vibration-damping capabilities of chlorobutyl elastomers.** Rodger N. Capps (Underwater Sound Reference Detachment, Naval Research Laboratory, P.O. Box 568337, Orlando, FL 32856-8337) and Linda L. Beumel (Texas Research Institute, 9063 Bee Caves Road, Austin, TX 78733-6201)

A chlorobutyl elastomer formulation has been developed at this laboratory (USRD) for potential vibration damping applications. It combines the attractive engineering properties of low compression set, good weather and ozone resistance, and good tensile and tear properties with desirable viscoelastic behavior for vibration control. An intensive investigation of the effects of plasticizers and different types and loadings of fillers upon physical properties and damping capability of the base polymer was conducted. Measurements of the dynamic Young's modulus and loss tangent over an extended frequency range were used in conjunction with the Ross-Ungar-Kerwin model [*Structural Damping*, edited by J. E. Ruzicka (American Society of Mechanical Engineers, New York, 1959), Sec. 3] to predict the damping of simple three-layer constrained structures. The model predictions were compared with actual damping measurements of composite plates. The observed differences between elastomers with different types of fillers are discussed in terms of mechanisms of interaction between the fillers and the gum phase of the elastomer. [Work supported by Code 1962 of David Taylor Research Center.]

## Session MM. Speech Communication VII: Cue-Equivalence and Perceptual Processing

Patricia K. Kuhl, Cochairman

*Department of Speech and Hearing Sciences, University of Washington, Seattle, Washington 98195*

Sigfrid D. Soli, Cochairman

*Biosciences Laboratory, 3M Center, St. Paul, Minnesota 55144*

## Contributed Papers

1:00

**MM1. Modeling listeners' perception of "silent center" syllables.** Terrance M. Nearey and Jean E. Andruski (Department of Linguistics, University of Alberta, Edmonton, Alberta T6G 2E7, Canada)

Recently, Verbrugge and Rakerd [*Language and Speech* 29, 39-57 (1986)] and Strange and Jenkins [*J. Acoust. Soc. Am. Suppl.* 1 82, S82 (1987)] have shown evidence that "silent center" CVC syllables (i.e., syllables wherein only the relatively rapid CV and transition portions are preserved, while the relatively steady-state midsections are excised) are well-perceived even when CV and VC portions from different speakers are combined ("hybrid syllables"). This study will present data from analogous experiments with Canadian English, but including both isolated vowels and /bVb/ syllables in an effort to test the degree to which consonantal coarticulation effects (as opposed to vowel-intrinsic properties) are responsible for the high intelligibility of hybrid syllables. The results obtained will be compared with predictions of the pattern recognition model of Nearey and Assmann [*J. Acoust. Soc. Am.* 80, 1297-1308 (1986)] for Canadian English vowels that exploits vowel-inherent spectral change.

1:15

**MM2. Cross-series selective adaptation with vowels.** Dawn L. Dutton and James R. Sawusch (Department of Psychology, SUNY at Buffalo, Amherst, NY 14260)

Cross-series selective adaptation has been found for vowels [J. J. Godfrey, *Percept. Psychophys.* 28, 103-111 (1980)], but the stimulus attributes relevant to adaptation were not clearly inferable. The predictions of two models of vowel recognition, the relational model and the octave band model [D. L. Dutton, J. R. Sawusch, and K. C. Tsoi, *J. Acoust. Soc. Am. Suppl.* 1 81, S18 (1987)] were tested using selective adaptation. An EH-AE series served as the base continuum. Five vowels were generated to serve as adaptors. The relationship between the test series and the adaptors could be represented in three ways, depending on the stimulus information assumed to be relevant for pattern recognition. The two models above employ different frequency-based representations. A third model assumes adaptation is based on the pattern of movement of spectral peaks over time. Adaptors were constructed to be similar to one endpoint of the test series based on each of these three representations. Results will be discussed in terms of what cross-series adaptation can tell us about the representations that may be used during vowel recognition. [Work supported by NINCDS.]

1:30

**MM3. Effects of nonstationary characteristics on vowel perception.** René Carré (Laboratoire de la Communication Parlée, ICP Unité Associée au CNRS, INPG-ENSERG, 46 Avenue Félix Viallet, Grenoble, France 38031)

Amplitude transition effects were studied by means of a formant synthesizer. The signal amplitude control is located either between the vocal source and the formant filters or between the formant filters and the output. In the first configuration, corresponding to the human vocal tract situation, at the beginning of the amplitude transition, an impulse response of the formant filters is obtained. In the second configuration, the formant filters are always in a stationary state and, hence, the spectrum structure of the formant filter outputs is harmonic. The distortions introduced by the amplitude transitions do not induce more information on the transfer function of the formant filters. Clear spectral differences were obtained when comparing the two synthetic signals, at their onsets. Information on formant frequency can be obtained in the case of the first configuration even if  $F_0$  is high. It has been shown that signal differences obtained by the two configurations are specifically processed by the human auditory system and induce displacements of [i]-[e] boundaries. Moreover, the vowel spectra corresponding to the boundaries do not have a harmonic component in the center of the first formant.

1:45

**MM4. Interpreting vowel "quality": The dimension of rounding.** Leigh Lisker (University of Pennsylvania, Philadelphia, PA 19104 and Haskins Laboratories, New Haven, CT 06511)

According to one view, speech perception depends crucially on the articulatory interpretation of the acoustic signal, since this would explain the identification of the acoustically different initial portions of two formant pseudospeech patterns such as, e.g., /di/ and /du/: Listeners "know" that their onsets are produced with very similar tongue gestures. Is this relation between perception and articulation versus acoustic signal typical, or is it instead restricted to certain phonetic units? For vowels it appears that different combinations of tongue and lip articulations yield very similar acoustic patterns and auditory qualities. The 18 "cardinal vowels" of Daniel Jones were presented to several groups of listeners asked to decide which had been produced with lip rounding. Error rates varied significantly over the vowel space, and showed a bias toward reporting front vowels as unrounded and back ones as rounded, a tendency that presumably reflects the relative frequencies across languages of front unrounded, back rounded, front rounded, and back unrounded vowels. Thus, if speech perception is crucially dependent on an articulatory interpretation of an acoustic signal, that interpretation need not be correct. [Work supported by NIH Grant HD-01994 to Haskins Laboratories.]

2:00

**MM5. More support for rate-based discrimination of second formant transitions.** M. J. Collins (Kresge Hearing Research Laboratory, LSU Medical Center, New Orleans, LA 70112 and LSU Communication Disorders, New Orleans, LA 70112), J. K. Cullen (Kresge Hearing Research Laboratory, LSU Medical Center, New Orleans, LA 70112, LSU Communication Disorders, New Orleans, LA 70112 and Department of Psychology, University of New Orleans, New Orleans, LA



70112), R. J. Porter (Kresge Hearing Research Laboratory, LSU Medical Center, New Orleans, LA 70112 and Department of Psychology, University of New Orleans, New Orleans, LA 70112), and D. F. Jackson (Kresge Hearing Research Laboratory, LSU Medical Center, New Orleans, LA 70112)

Two prior studies [Besing *et al.*, *J. Acoust. Soc. Am.* **81**, 535 (1987); Porter *et al.*, *J. Acoust. Soc. Am. Suppl.* **1** **82**, S81 (1987)] investigated differential thresholds for onset frequency of second formants (30-ms transitions) and for transition duration (60- and 120-ms transitions). The extent of both rising and falling (to 1800-Hz) transition was systematically varied to yield differing rates-of-change for stimuli. Transition discrimination was examined for formants isolated and in a first formant context. The question addressed was whether discrimination varied as a function of the simple stimulus dimensions of frequency and time or in relation to the more complex domain of the rate-of-change of frequency over time. Several aspects of the results supported a rate-of-change interpretation. The present study further tests that interpretation by extending observations to discrimination of onset frequency for 60- and 120-ms transitions. Results for six listeners indicate that onset-frequency discrimination of 60-ms transitions is also based on rate-of-frequency change as was the case for 30-ms transitions; discrimination of 120-ms transitions may be, in part, frequency based. In addition, rate-based discrimination appears to be nonlinear in that higher rates-of-change yield smaller difference limens relative to a reference than do lower rates-of-change. [Supported by NIH-NINCDS and the Louisiana Lions Eye Foundation.]

2:15

**MM6. Rate normalization for the /ba-wa/ contrast.** Rebecca E. Eilers, D. Kimbrough Oller, and Rebecca Burns (Departments of Psychology and Pediatrics, P.O. Box 016820, University of Miami, Miami, FL 33101)

A controversial hypothesis in speech perception posits an important role for rate context in determining the boundary location between /ba/ and /wa/. In general, Miller and Liberman (1979) suggest that longer formant transitions are necessary to cue /wa/ when these syllables contain long rather than short vowels while Shinn *et al.* (1985) suggests that boundary location is influenced by rate context only under special circumstances. The present work reexamines the rate context hypothesis using four synthetic /ba/ to /wa/ continua differing in vowel duration and four continua varying in both duration and auxiliary synthesis parameters. Adult subjects participated in several standard and nonstandard perception tasks including, identification, ABX discrimination, same-different discrimination, and VRISD, a repeating background detection task. Results indicate a small rate effect that may be largely accounted for by the shortest vowel durations. Examination of individual subject data across tasks suggests that the effect may not have important linguistic ramifications. [Work supported by NIH # HD21534.]

2:30

**MM7. The role of rate of transition in the perception of place of articulation.** Renee A. E. Zakia and John Kingston (Department of Modern Languages and Linguistics, Morrill Hall, Cornell University, Ithaca, NY 14853-4701)

Kewley-Port *et al.* ["Perception of static and dynamic acoustic cues to place of articulation in initial stop consonants," *J. Acoust. Soc. Am.* **73**, 1779-1783 (1983)] showed that listeners can reliably identify the place of articulation of initial bilabial and alveolar stops in stop-vowel syllables with severely truncated vowels from just the first 20 ms of the syllable, which includes the burst and the beginning of the formant transitions. However, they showed that listeners must hear substantially more of the formant transitions, 40 ms or more, to identify the stop's place of articulation when the stop is velar. They speculate that their listeners' need for a longer transition may reflect a slower change in formant frequencies observed overall for velars compared to stops at other places of articulation. Slower formant frequency changes for velars probably arise from the greater inertia of the tongue body compared to the tongue tip or lip, as do the longer voice-onset-times (VOTs) observed for velars versus bilabials or alveolars. If a listener has other (spectral) cues that a stop is a velar, then he will accept stimuli with longer VOTs as voiced than if the other cues indicate another place of articulation. Though the longer VOTs are not themselves cues that the stop is a velar, they may enhance that percept [cf. K. Stevens *et al.* "Toward a phonetic and phonological theory of redundant features," in *Invariance and Variability in Speech Processes*, edited by J. Perkell and D. Klatt (Erlbaum, Hillsdale, NJ, 1986), pp. 426-449]. Again assuming that other cues to place are present, slower transitions may enhance the identification of a stop as velar in a similar way, but could have either a negligible or deleterious effect on identifying a stop as bilabial or alveolar. In this study, ABX-format discrimination experiments are being conducted in which listeners hear short (100 ms) CV syllables in which the second and third formant transitions match natural values for the syllables [ba, da, ga, bi, di, gi] in direction and magnitude, while their rate of change is varied in 15-ms steps across a range of 15-90 ms. The stop bursts have been excised from these stimuli so the only cues to place are in the formant transitions. It is expected that listeners' ability to discriminate place of articulation to degrade in two directions, with artificially short transitions for velars and artificially long transitions for nonvelars.

2:45

**MM8. An evaluation of psychophysical trading relations and perceptual cues.** Richard E. Pastore and Jody Kaplan Layer (Department of Psychology, SUNY at Binghamton, Binghamton, NY 13901)

Trading relations have been the empirical and theoretical focus of a number of research studies over the last decade. This paper provides a careful examination of the concept of a cue, the concept of a trading relation between cues for a given perceptual contrast, and published claims for empirical demonstrations of trading relations. While true trading relations should be expected for most types of perceptual phenomena, it was discovered that many studies claiming to find a trading relation have only demonstrated the effects of a single cue. Furthermore, some aspects of trading relations may be based upon a failure to adequately consider critical underlying assumptions about the nature of the continuum along which the dependent variable is defined. The theoretical implications of trading relations will be discussed. [Research supported in part by a NSF grant to the first author.]

3:00-3:15

Break

3:15

**MM9. Integrality and trading relations across syllabic and lexical boundaries.** Gail R. Tomiak and James R. Sawusch (Department of Psychology, Park Hall, State University of New York at Buffalo, Amherst, NY 14260)

Previous research using a speeded classification task with noise-tone analogs of monosyllables and disyllables has shown that monosyllabic stimuli show integral processing of adjacent phones when perceived as speech, but separable processing of adjacent pitch and amplitude information when heard as nonspeech. Furthermore, this integrality effect was found to be more pronounced within syllables than across syllable boundaries. It was suggested that this pattern of results reflects a speech mode of processing. The purpose of the present experiment was to determine whether trading relations can also be attributed to language specific coding processes or, to the auditory integration of information spanning syllable (and lexical) boundaries. Analogs of speech stimuli known to produce a trading relation [Repp *et al.*, *J. Exp. Psychol.: Hum. Percept. Perform.* 4, 621-637 (1978)] were constructed and tested in a speeded classification task. The results, and implications of these findings, will be discussed. [Work supported by NINCDS.]

3:30

**MM10. Trading relations in intonation.** Gerald McRoberts and Donald Shankweiler (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Department of Psychology, University of Connecticut, Storrs, CT 06268)

Trading relations in the perception of speech have been reported both for segmental aspects of speech [Best *et al.*, *Percept. Psychophys.* 29, 191-211 (1981)] and for intonation, a suprasegmental aspect of speech [Studdert-Kennedy and Hadding, *Lang. Speech* 16, 293-313 (1973)]. However, trading relations in speech production have not been reported. Two studies demonstrate this phenomenon in the production of intonation. In experiment 1, male speakers produced two sentences ("November" and "I did it") both as questions and as statements. For questions, a trading relation between the peak  $F_0$  and the amount of  $F_0$  rise on the final syllable was found: As peak  $F_0$  increased, the amount of  $F_0$  rise on the final syllable decreased. This result is consistent with the findings in perception and suggests a link between the perception and production of intonation. In experiment 2, male actors produced questions and statements using the same two sentences. Peak  $F_0$  was varied in two ways: linguistically (emphatic/nonemphatic stress), or affectively (simulated happy and sad emotions). Utterance types in each condition were verified by listeners. The existence of a trading relation between peak  $F_0$  and  $F_0$  final rise for questions was confirmed for the emphatic stress condition, but not for the affective condition. This dissociation confirms indications from perceptual studies of a linguistic origin for the effect. These results demonstrate a perception-production link for linguistically relevant uses of intonation. [Work supported by NIH #NS-24655 and #HD-1994.]

3:45

**MM11. Perceptual effects of attention to phonetic versus "musical" properties of sinewave speech.** Catherine T. Best (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511 and Department of Psychology, Wesleyan University, Middletown, CT 06457), Michael Studdert-Kennedy (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511, Department of Linguistics, Yale University, New Haven, CT 06520, and Department of Psychology, University of Connecticut, Storrs, CT 06268), and Sharon Manuel (MIT Speech Laboratory, MIT, Cambridge, MA 02139)

A continuing debate in speech perception research has been whether phenomena such as categorical perception and trading relations are explained by basic psychoacoustic mechanisms or rather by a specialized phonetic mode of perception. Sinewave speech analogs have provided an opportunity to test these alternatives, since the same stimuli can be heard as either distorted speech or nonspeech sounds by different groups of listeners. The psychoacoustic hypothesis would predict similar between-group performance, while the phonetic hypothesis would predict system-

atic group differences. Several recent findings have favored a distinct phonetic mode in perception of sinewave speech. Best and Studdert-Kennedy [Proc. 10th Int. Congr. Phon. Sci (1983)] used a labeling test with a sinewave-analog /ra/-/la/ continuum, and found that speech-biased listeners performed in a clearly categorical manner, whereas "music"-biased listeners performed near chance. The present research strengthened that finding, by employing AXB identification *without* overt labels, and AXB discrimination. Control tests with a full-formant /ra/-/la/ continuum and sinewave analogs of their isolated F3 trajectories were included. The two groups performed identically with each control continuum, yet they differed dramatically with the sinewave syllables. In support of the phonetic hypothesis, only the speech-biased listeners systematically classified the stimuli into two clear categories, and showed improved discrimination around the category boundary. [Work supported by NIH grant HD-01994 to Haskins Laboratories.]

4:00

**MM12. Can simultaneous presentation of print influence the detection of speech in noise?** Ram Frost and Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

When spoken words are masked by noise having the same amplitude envelope, subjects report they hear the word much more clearly when they see its printed version at the same time. Using signal detection methodology, it was investigated whether this subjective impression reflects a change in perceptual sensitivity or in bias. Signal-plus-noise and noise-only trials were accompanied by matching print, nonmatching (but structurally similar) print, or a neutral visual stimulus. These results revealed a strong bias effect: The matching visual input apparently made the masking noise sound more speechlike, but it did not improve the detectability of the speech. (However, reaction times of correct detections were reliably shorter in the matching condition, suggesting perhaps a subliminal facilitation.) The bias effect was much smaller when nonwords were substituted for the words. Thus it seems that subjects automatically detect correspondences between speech amplitude envelopes and printed stimuli, and they do this more efficiently when the printed stimuli are real words. This supports the hypothesis, much discussed in the reading literature, that printed words are immediately translated into an internal representation having speechlike characteristics. [Work supported by NICHD.]

4:15

**MM13. The interaction of visual place and auditory voicing information during the perception of speech.** Kerry P. Green and Patricia K. Kuhl (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

Previous studies using speeded classification paradigms have been used to test whether the information that underlies the perception of place and voicing is processed in an integral as opposed to a separate fashion. Results of auditory only experiments show that reaction times (RTs) for place classification in a condition in which stimuli vary along both place and voicing (the orthogonal condition) are longer than RTs in a control condition in which stimuli vary only in place. These data are taken in support of the idea that the information underlying place and voicing are integrally processed. This study was concerned with whether or not place classification would demonstrate the same pattern of results when the place feature was determined by both auditory and visual (the face of the talker) information. To do this, the auditory tokens /ibi/ and /ipi/ were paired with a video display of a talker saying /ibi/ or /igi/, and presented to subjects for speeded classification of place or voicing. Because of the "McGurk effect," these stimuli were perceived as /ibi, ipi, idi, or iti/. The results showed that RTs for classifying phonetic features in the orthogonal condition were significantly longer than in the control condition. These results were obtained for both place and voicing judgments. Thus auditory-visual results appear to mirror those obtained in previous studies using only auditory stimuli. [Work supported by NIH.]

**MM14. Factors affecting the integration of auditory and visual information in speech: The level effect.** Patricia K. Kuhl, Kerry P. Green (Department of Speech and Hearing Sciences), and Andrew N. Meltzoff (Department of Psychology, University of Washington, Seattle, WA 98195)

The "McGurk effect" is a phenomenon in which an illusory syllable is perceived when discrepant auditory and visual information are combined. For example, an illusory /da/ is perceived when an auditory syllable /ba/ is paired with a video display of a talker saying /ga/. In a series of studies reported here, it was found that the level of the auditory signal substantially affects the illusion. Moreover, this effect of auditory level works in a direction that is counterintuitive. One might have postulated that as the level of the auditory signal is increased, making the auditory signal more pronounced, the illusion would decrease. These results indicate just the opposite. As the level of the auditory signal goes up, rather than it outweighing the visual signal and thereby decreasing the illusion, the effect is to increase the number of illusory responses. Paired were the same auditory syllable /ba/ at three levels, soft (45 dB SPL), moderate (58 dB SPL), and loud (66 dB SPL), with a video display of /ga/. Significant increases in illusory /da/ responses occurred as the auditory signal increased from soft to moderate and again from moderate to loud. The work thus shows that auditory level is an important determinant in the perception of the illusion. The paradox of the situation is that as the auditory signal increases in strength, the resulting percept becomes less veridical rather than more so. [Work supported by NIH.]

**MM15. Further studies on the auditory-visual articulation index.** Ken W. Grant (Research Laboratory of Electronics, MIT, Cambridge, MA 02139)

Previous work on the auditory-visual articulation index [Ken W. Grant, *J. Acoust. Soc. Am. Suppl. 1* 79, S4 (1987)] showed that filtered bands of speech with different center frequencies but roughly the same auditory intelligibility in quiet also have the same auditory-visual intelli-

gibility. Furthermore, it was shown that these results could be fairly well predicted by articulation theory. The present study extends this work by examining the theory's ability to predict auditory-visual intelligibility for filtered bands of speech in noise as well as for various combinations of nonadjacent 1/3 octave bands in quiet. A comparison of predicted and obtained intelligibility results will be discussed for each filtered band as a function of speech-to-noise ratio. In addition, the assumption that the auditory AI obtained for each separate band can be added to predict the auditory and auditory-visual AI when the bands are combined will be tested. [Work supported by NIH.]

**MM16. Choosing an appropriate technique for measuring the benefit of lipreading to speech perception in noise.** Quentin Summerfield, Alison MacLeod, Diana Field, and John Foster (MRC Institute of Hearing Research, University Park, Nottingham NG7 2RD, England)

The reliability of two procedures for measuring the benefit provided by lipreading and speech perception in noise were compared. Benefit was measured as the difference between two signal-to-noise ratios (SNRs) giving a criterion level of intelligibility against a fixed level of white noise when speech was presented: (i) in audio-alone and (ii) in audio-visual conditions. In the first technique, sentences were presented using an up-down rule to converge on the SNR giving 50% intelligibility. In the second technique, continuous speech was presented and subjects adjusted the SNR until they could "just understand every word that was spoken." For the same duration of testing, the sentence-based procedure yielded an estimate of benefit that differed more reliably from zero and that had a greater test-retest reliability. This estimate also correlated more highly with an independent estimate of a subject's ability to lipread sentences with vision alone, probably because the sentence-based procedure places a higher premium on the use of visual cues. Overall, the sentence-based procedure is the better measurement tool, although the self-adjustment procedure may have some value in rehabilitation.

THURSDAY AFTERNOON, 19 MAY 1988

WEST BALLROOM B, 1:20 TO 2:55 P.M.

### Session NN. Underwater Acoustics VI: Underwater Noise

David J. Thomson, Chairman

*Defence Research Establishment Pacific, FMO Victoria, British Columbia V8W 2Y2, Canada*

Chairman's Introduction—1:20

#### Contributed Papers

**NN1. Topographic noise stripping at oblique angles over continental slopes.** Donald R. Del Balzo, Deborah L. Head, and Mona J. Authement (Naval Research and Development Activity, NSTL, MS 39529)

In a previous presentation [D. R. Del Balzo, M. J. Authement, and C. T. Mire, *J. Acoust. Soc. Am. Suppl. 1* 78, S3 (1985)], the effect of a 2.8-deg slope on the noise field generated by 100-Hz sources was discussed.

The sources were located in the deep-water basin and over the shallow-water shelf. Propagation for a summer Sargasso sound-speed profile was considered for sources at various ranges from the coastline and located in the range-depth plane normal to the coastline. The presence of a noise notch, a region of relatively low noise, at shallow depths over the slope, was discovered. In this presentation, the analysis will be expanded to include seasonal effects (summer and winter sound-speed profiles), frequency dependence (30 and 100 Hz), and sources located at oblique angles out of the range-depth plane normal to the coastline. The effect of these parameters on the noise field and noise notch will be discussed.

1:40

**NN2. Temporal and spatial correlations for low-frequency ambient noise in a diffuse ice edge.** N. C. Makris (Department of Ocean Engineering, MIT, Cambridge, MA 02139)

Low-frequency ambient noise data acquired during Mizex 84 have been analyzed for temporal and spatial variations. Several concurrent time series of rms pressure in the 25- to 50-Hz band sampled at 30-s intervals and of a few hours in duration were constructed from hydrophone data taken at various separations within a 10-km range. During noise episodes, a time scale on the order of 10 min has been obtained via autocorrelations. Also, variations in noise level of up to 7 dB between sensors as close as 1/2 km, together with crosscorrelations which are high at separations of 5 km but significantly smaller at half that distance, suggest a spatially banded or striated noise generation process. [Work supported by the Office of Naval Research.]

1:55

**NN3. Analysis of marginal ice zone noise events.** Chi-Fang Chen (Department of Ocean Engineering, MIT, Cambridge, MA 02139)

Underwater ambient noise data from MIZEX 84 were examined and transient noise events were detected and studied. Event locations were determined and their peak values used to calculate the source strength based on a simple vertical dipole model. The results indicate that the source strength is about the same order of magnitude as the strength of sources at corresponding frequency in the central Arctic. Event signatures were classified, but, in contrast to the central Arctic case, are considerably longer than their scaled counterparts. [Work supported by Office of Naval Research.]

2:10

**NN4. Statistical characterization of Arctic under-ice acoustic bursts and tonals.** Samuel M. Nagle, Clinton T. Siedenburg (Honeywell, MSD, 6500 Harbour Heights Parkway, Everett, WA 98204-8899), and James A. Ritcey (Department of Electrical Engineering, FT-10, University of Washington, Seattle, WA 98195)

A 10-min time series of FRAM II Arctic under-ice noise data is analyzed, extending the results of an earlier study [J. G. Veitch, A. R. Wilks, and S. C. Schwartz, "Characterization of Arctic undersea noise," pp. 45, Dep. Stat., Princeton University (June 1983), AD-A130 397/3 (No. 22, 1983)]. Specific analysis is conducted on the impulsive noise bursts and tonals that are characteristic of underwater noise when an ice cover is present. Point process modeling of the times of arrival of the impulse bursts is performed by testing the empirical distribution of the burst event interarrival times against an exponential parent distribution, characteristic of a Poisson process. Goodness-of-fit tests are invoked, as well as a test

for serial correlation. Results suggest that the burst event point process may be modeled as a piecewise constant nonhomogeneous Poisson process. This model is accepted at >5% confidence for all tests applied. Though a limited data set is utilized, the analysis methodology developed is applicable to larger sets of data. Spectral content of tonal components are also examined using Thompson's multiple window technique, and a harmonic relationship is found to exist.

2:25

**NN5. Distribution of surface-generated noise in a three-dimensional environment.** John S. Perkins, W. A. Kuperman, F. Ingenito (Naval Research Laboratory, Washington, DC 20375), and Michael P. Porter (SACLANT Undersea Centre, I-19026, La Spezia, Italy)

Modeling the distribution of noise generated at the ocean surface over a wide ocean area with a spatially variable source level distribution in a three-dimensional (3D) environment requires the capability of efficiently computing the acoustic field from all points on the surface to all points in the ocean. A previously developed noise theory is modified [W. A. Kuperman and F. Ingenito *J. Acoust. Soc. Am.* **67**, 1988-1996 (1985)] and is combined with an approximate numerical technique for computing transmission loss in a 3-D environment based on adiabatic mode theory [M. P. Porter, W. A. Kuperman, F. Ingenito, and A. Piacsek, *J. Acoust. Soc. Am. Suppl.* **1 81**, S9 (1987)]. The wave equation solution of the acoustic field (or its cross-spectral density) from any point to any point (or to any pair of points) reduces to a "spreadsheet" type manipulation of precomputed modal solutions. This 3-D range-dependent numerical scheme is shown to reduce to analytic expressions for the distribution of noise in a horizontally stratified ocean.

2:40

**NN6. An analysis of low-frequency wind-generated ambient noise source levels in the ocean.** W. M. Carey, D. J. Kewley,<sup>a)</sup> D. G. Browning, and W. A. VonWinkle (New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

Historically, the greatest amount of published ambient noise data has been obtained in Northern Hemisphere Oceans. However, principally due to ship noise contamination and lack of directionality, only a limited portion can be used for determining low-frequency and wind-generated ambient noise source levels. Some data have been extracted and combined with the Southern Hemisphere database [A. S. Burgess and D. J. Kewley, *J. Acoust. Soc. Am.* **73**, 201-210 (1983)]. These resulting wind-generated ambient noise source levels are difficult to compare due to differences in the source level definition and descriptions by several investigators. A standard set of source levels is proposed based on a monopole beneath the pressure release surface and an equivalent dipole at the surface. The ambient noise source level data reported in the literature are shown to be in general agreement. <sup>a)</sup> Present address: Weapons Systems Research Laboratory, DSTO, GPO Box 2151, Adelaide, S.A., 5001, Australia.

## Session OO. Physical Acoustics VI and Bioresponse to Vibration III: Extracorporeal Shock Wave Lithotripsy—Biological Aspects

David T. Blackstock, Chairman

*Applied Research Laboratory, University of Texas, P. O. Box 8029, Austin, Texas 78713-8029*

Chairman's Introduction—1:25

### Invited Papers

1:30

**OO1. Cavitation in lithotripsy.** Edwin L. Carstensen (Departments of Electrical Engineering, Biophysics and the Rochester Center for Biomedical Ultrasound, University of Rochester, Rochester, NY 14627)

The pressure amplitudes of the shock waves used in lithotripsy far exceed thresholds for transient cavitation. Where appropriate nuclei exist within the body, it is highly probable that bubbles will form and collapse violently, giving rise to potentially undesirable side effects of the treatment. Evidence from basic and clinical studies in support of this conclusion is accumulating rapidly. *Drosophila* larvae have served as useful biological models for studies of cavitation-related phenomena because of the highly reproducible microbubbles within their respiratory system. Using approximately 100 000, 1- $\mu$ s pulses from a 2-MHz piezoelectric source, the threshold for killing of larvae occurs at a pressure amplitude of  $\sim 0.7$  MPa. With an electrohydraulic (unfocused spark source) lithotripter, three positive spikes of  $\sim 1$  MPa are sufficient to kill roughly one-half of the exposed larvae. This is two orders of magnitude less than the largest reported pressures for clinical lithotripters.

2:05

**OO2. Biological effects of shock waves.** M. Delius (Institute for Surgical Research, University of Munich, Munich, Federal Republic of Germany), H. Eizenhoefer, R. Denk (Dornier Medizintechnik GmbH, Gerning, Federal Republic of Germany), H. Liebich (Veterinary Anatomy, University of Munich, Munich, Federal Republic of Germany), and W. Brendel (Institute for Surgical Research, University of Munich, Munich, Federal Republic of Germany)

The effect of shock waves on normal canine kidneys was examined in two groups of dogs whose right kidneys were exposed to 3000 shock waves in a Dornier HM II lithotripter. The groups differed only in the rate of shock wave administration, which was 100 and 1 per second, respectively. At autopsy, more hemorrhages and hemolysis occurred if shock waves were administered at a rate of 100 per second. In another experiment, extracorporeal shock waves were slowly administered to piglet liver. They induced two types of changes as detected by conventional real-time ultrasound: Transient changes consisted of bursts of bubbles in liver veins that were flushed away with the blood flow. Permanent changes, which appeared later after several hundred shock waves, consisted of a bright area of increased air content in the high-pressure field of the wave. This is evidence that shock waves generate cavitation *in vivo*. Supported by further experiments, a model of shock wave action is proposed: The negative pressure part of the shock wave generates microbubbles which, if encountered by the positive pressure part of the following shock wave, generate the shock wave effects. [Work supported, in part, by Kurt Körber Stiftung.]

2:40

**OO3. Materials fragmentation by shock waves.** B. Finlayson (Division of Urology, Box J-247 JHMHC, University of Florida, Gainesville, FL 32610), Mohamed Nasr (11 Medan El-Tahreer No. 22, Cairo, Egypt), and J. Paul Whelan (29 Heritage Drive, #19, Stoney Creek, Ontario L8G 4T4, Canada)

Although extracorporeal shock wave lithotripsy (ESWL) is an important technique, the details of how it works—how shock waves interact with stones to cause disruption—and how to optimize its effect are not known with certainty by practitioners of the art. Three position-dependent mechanisms of disruption were observed. Position dependence was probably due to focusing of the shock wave by an ellipsoidal reflector. Near the focus, the front surfaces of the test objects were roughed (spalated), probably by collapse of cavitation bubbles caused by passage of the shock wave. Also, midspecimen cleavage was seen, presumably caused by back wall reflection tensile stress. Distal to the focus back wall, spallation occurred, possibly due to tension caused by the expanding shock wave. Sieving fraction observations made of the effect of shock waves on "z brick" show that fragmentation has an approximate first-order dependence on the number of shocks and a second-order dependence on the voltage at which the shocks were generated. The multiphase character of the z brick was distinctly revealed in plots of percent fragmentation versus shock number; z brick is not a satisfactory test material, and there is a need for a well-characterized set of test materials.

**OO4. Extracorporeal shock wave lithotripsy of gallstones: Objectives, current limitations, and preliminary *in vitro* and *in vivo* observations.** J. L. Thistle and B. T. Petersen (Division of Gastroenterology, Mayo Clinic, Rochester, MN 55905)

Surgical removal of gallstones is performed about 500 000 times per year in the United States. Although the usual cost ranges from \$5000 to more than \$10 000 per patient, the operative mortality is less than 1% for otherwise healthy persons. Nonsurgical dissolution of gallstones is often possible, but many gallstones contain calcium compounds that slow or prevent dissolution using the direct content cholesterol solvent, methyl tert-butyl ether (MTBE). Extracorporeal shock wave lithotripsy (ESL) has potential clinical utility for facilitating dissolution and possibly allowing spontaneous or induced passage of small fragments via the bile ducts and intestines. Obstacles to be overcome include predictable fragmentation of multiple stones up to 3 cm in diameter so that all particles are small enough to be rapidly dissolved or pass safely through the bile ducts ( $<3$  mm). This must be achieved using biologically tolerable shock wave dosages without requiring general anesthesia within one or two treatment sessions. Acute and chronic safety, patient acceptance, and cost must be competitive with surgical treatment. Initial experience suggests that only solitary noncalcified stones (10% of patients with stones) have a high probability of success using ESL plus oral bile acid dissolution therapy. Improving ESL technology and direct contact fragment dissolution using MTBE should increase the clinical utility of shock wave lithotripsy of gallstones.

### Contributed Papers

3:50

**OO5. Cavitation in flowing media by lithotripter shock waves *in vitro* and *in vivo*.** Douglas L. Miller (Battelle Northwest, P.O. Box 999, Richland, WA 99352), M. Delius (Institute for Surgical Research, University of Munich, Munich, Federal Republic of Germany), A. R. Williams (Department of Medical Biophysics, University of Manchester, Manchester, United Kingdom), and W. Schwarze (Dornier-Medizintechnik, Munich, Federal Republic of Germany)

Cavitation produced by lithotripter shock waves was characterized *in vitro* in water and blood, and *in vivo* in aortic blood by means of a resonant bubble detector. The 1.6-MHz detector can detect and count  $4 \pm 1$ - $\mu$ m-diam bubbles flowing through it by receiving their second harmonic emissions at 3.2 MHz. Spark-gap lithotripters were used to expose the flowing liquid upstream of the detector at the shock wave focus. This system was readily able to detect bubbles resulting from shock wave-induced cavitation in both water and blood flowing through plastic tubes *in vitro*, and even in blood pumped by the heart through a plastic arteriovenous shunt. Multiple (up to several hundred) bubble counts were obtained for each shock wave. However, for 200–400 shock wave exposures of each of two dogs, this system was unable to detect evidence of shock wave-induced cavitation activity occurring within the intact vascular system *in vivo*. [Work supported by Dornier Medizintechnik and by NIH Grant CA 42947.]

4:10

**OO6. Cavitation processes induced by weak shock waves propagating in tissue.** H. Koch, M. Grünwald, and H. Hermeking (Dornier Medizintechnik GmbH, Industriestr. 15, 8034 Germering, West Germany)

Human application of single-cycle focused shock waves requires an intense study of the interaction processes associated with shock wave propagation in biological media. In this connection, shock wave-induced cavitation processes on the scale of a single biological cell are of fundamental importance. On the basis of a model analysis including different propagation media for the bubble–shock wave interaction and various shock wave profiles, possible hazards are discussed and compared with experimental and clinical results.

4:30

**OO7. Determination of design parameters for an ultrasonic kidney stone disintegrator.** R. Agarwal and V. R. Singh (Department of Instrumentation and Biomedical Ultrasonics, National Physical Laboratory, New Delhi, 110012, India)

Acoustic techniques are now increasingly used for the removal of kidney stones without surgery. In the past, generally, acoustic shock wave techniques and lithotripsy have been used. However, focused ultrasound may be used to disintegrate such stones, without any effect on the surrounding tissues. Ultrasonic characteristics of kidney stones are studied to determine further the design parameters such as modulus of elasticity, acoustic impedance, frequency of vibrations, etc., to develop a kidney stone disintegrator. The value of ultrasonic velocity in kidney stone *in vitro* is found to be almost double that of water or soft biological tissues.

4:50–5:20

### Bull Session

## Session PP. Musical Acoustics IV: Arthur H. Benade Memorial Session II: Horns, Strings, and Harmony

Thomas D. Rossing, Chairman

*Department of Physics, Northern Illinois University, DeKalb, Illinois 60115*

Chairman's Introduction—1:30

*Invited Papers*

1:35

**PP1. Reflections on the development of a physicist.** R. E. Chrien (Brookhaven National Laboratory, Physics Department, Upton, NY 11973)

We are here to honor the contributions of Arthur Benade to understanding the physics of musical instruments. These unique contributions were brought to fruition by a set of fortunate coincidences that resulted in a shift from Benade's original career orientation in nuclear physics to one of musical acoustics. As Benade's first student, from 1952 to 1957, I was in a position to view this change at first hand, and perhaps to understand how it occurred better than most others. The coming together of Arthur's deep interest in pedagogy, his broad view of the unity of physics, and the conducive atmosphere present in the department at Case—with its historical ties to acoustics—were responsible for initiating his career in the acoustics of musical instruments. Lessons for us all may be drawn from his ability to combine his vocation and his avocation in a harmonious and productive manner. I would like to share with you my remembrances of this formative time in his professional life.

2:05

**PP2. A simplified model for a beating reed—Preliminary results.** W. E. Worman (Department of Physics, Moorhead State University, Moorhead, MN 56560) and A. H. Benade (Physics Department, Case Western Reserve University, Cleveland, OH 44106)

This paper presents early results of a model for the behavior of a musical wind instrument with a beating reed in the case where the air flow is completely blocked for a finite part of each period. The parameters used are appropriate for a clarinet. The method is as follows: (a) Assume a waveform for the transaural pressure,  $p_{in}$  (pressure difference across the reed); (b) calculate the reed position assuming that the reed is a simple harmonic oscillator driven by  $p_{in}$ ; (c) use the pressure and reed position to calculate the flow into the mouthpiece; (d) use the flow and input impedance of the air column to calculate the transaural pressure,  $p_{out}$ ; and (e) adjust the assumed  $p_{in}$  so that the calculated  $p_{out}$  agrees with it. The results obtained so far allow some general conclusions. The pressure waveform is "squarish," almost symmetrical, and varies between about zero and twice the blowing pressure. The behavior is independent of system parameters over wide limits and is controlled by energy exchanges that occur when the transaural pressure is near its minimum.

2:35

**PP3. Recollections of studies in the laboratory of A. H. Benade: 1975–1978.** Stephen C. Thompson (Acoustics and Sensor Systems, Ocean Systems Division, Gould Inc., 18901 Euclid Avenue, Cleveland, OH 44117)

From 1975 to 1978 the author had the honor of studying acoustics as a graduate student in the laboratory of Professor A. H. Benade. By this time, much of the fundamental work in understanding the physics of wind instrument air columns, tone holes, and the nonlinear interaction between the air column and its excitation mechanism had been developed. *Fundamentals of Musical Acoustics* had just been sent to the publisher. Questions concerning some of the more subtle aspects of wind instrument behavior were beginning to be asked: reed resonance effects, spectral transformations from the instrument to the room, mutual interaction of the radiation from adjacent tone holes, and the modification of air column resonances by the player's airway impedance. And all of the new physical understanding was routinely being converted to the musical instrument craftsman's intuitive methods for improving the playing behavior of instruments. For Art, the theoretical understanding of physical phenomena and the practical application of these new ideas in the "real world" of the musician held equal status. This constant interplay of theory, experiment, and practical application created a chain reaction of ideas which is still underway. This paper will recount the author's recollections of the atmosphere and developments of that time period.

**PP4. Woodwind design algorithms to achieve desired tuning.** Douglas H. Keefe (Systematic Musicology Program, School of Music, DN-10, University of Washington, Seattle, WA 98195)

Techniques exist to analyze woodwinds of known geometry. The inverse problem is a technique by synthesis which determines tone hole placements to approximate ideal tuning for tones the instrument is intended to produce. Two algorithms using a relaxation technique have as inputs: mouthpiece equivalent length, bore shape, tone hole dimensions, and mode number. Outputs include the tone hole positions, deviation from harmonic alignment of modal resonances, and open hole lattice cutoff frequencies. Bad designs are detected, forcing the designer to modify the bore or tone hole geometry. The first algorithm, due to Benade, uses a model based upon regular, semiinfinite tone hole lattices, and he employed it in designing the NX clarinet and other woodwinds. The new algorithm incorporates damping, a general tone hole model, lattice irregularities and finite bell termination. The algorithm allows calculation of input impedance and reflection functions for subsequent time domain simulations. The playing behavior of flutes constructed using these algorithms correlates well with predictions. Comparisons of the algorithms and their sensitivity to variations in tone hole impedance models will be discussed.

3:35

**PP5. From musician to listener via first principles and careful observation.** Peter L. Hoekje (Department of Medicine, Case Western Reserve University, Cleveland, OH 44106)

Many of the elements comprising the acoustics of musical wind instruments have been known since the time of Lord Rayleigh and Helmholtz. Yet, only in the last few decades have the contributions of acousticians been able to weigh significantly with those of artisans (whose names like Boehm, Conn, Hotteterre, or Sax have become familiar to us through the instrument-making dynasties founded on their acoustic insights) in the creation of fine instruments. Art Benade was a leader in these recent successes. Part of the legacy he leaves us is a methodology for design (or redesign) of wind instruments. It describes the adjustment of tone holes, bore shapes, and reeds in order to improve mode cooperation and decrease energy loss, verified by a series of simple playing experiments. By his unique insight, theoretical results arising from fundamental physical principles are identified with common attributes of "good" instruments. By similar study, he improved the understanding of the perception of sound in rooms. The life work of Art Benade was the isolation of factors of importance to musicians by sound scientific practice, inspired observation, and careful experimentation.

4:05

**PP6. A. H. Benade and the perception of music.** Ian M. Lindevald (Institute of Electroacoustics, Technical University of Munich, Arcistrasse 21, D-8000 München 2, West Germany)

Benade's fascination with the great skill by which the auditory system understands the complicated signals supplied to it grew from his primary interest in the physics of musical instruments. Often, solutions to problems of physics would force questions of perception. The link between physics and perception was natural for him, rounding out his knowledge of musical instruments as sound sources and the concert hall as a transmission system with serious inquiry regarding the auditory system as the sound's receiver. Benade's work in musical perception was guided by his devotion to "real world" acoustics and reflected his commitment to approaching his science from the viewpoint of those most affected by music, the musicians and experienced listeners. The author, who was Benade's last degree-earning doctoral student, will discuss the work in music perception: from the coffee room queries, through the formal research, and, finally, on to the continuing efforts to answer the questions that Benade left, so that we too may have some fun. [Work supported by the ASA's F. V. Hunt Postdoctoral Fellowship.]

### *Contributed Paper*

4:35

**PP7. Some memories of A. H. Benade.** Zili Li (Department of Physics, Case Western Reserve University, Cleveland, OH 44106)

Dr. Benade was my thesis advisor after I got my candidacy for Ph.D. in May 1985 and we have worked together since then. Art's rigorous

attitude toward science, profound knowledge of physics and acoustics, unique ability doing both theory and experiment, and the combination of humorous languages and pleasant personality gave me an unforgettable impression, especially for a student from China. I'll recall some of my experiences working with him, starting with my first impression of him to the last days we spent together in our laboratory.



## Session QQ. Noise V: Noise Measurement and Propagation

Govindappa Krishnappa, Chairman

*Engineering Lab. M7, National Research Council of Canada, Montreal Road, Ottawa, Ontario K1A 0R6, Canada*

Chairman's Introduction—1:30

## Contributed Papers

1:35

**QQ1. Scattering/diffraction effects of a sound intensity probe incorporating direct particle velocity measurements.** G. Krishnappa (Division of Mechanical Engineering, National Research Council, Ottawa, Ontario K1A 0R6, Canada)

Experimental studies were carried out to examine the scattering/diffraction effects of a sound intensity probe that determines particle velocity directly by using two pairs of ultrasonic transmitters and receivers. The measurement techniques employed were to detect the change in the separation distance acoustically of two closely spaced microphones placed in the immediate vicinity of the velocity measuring probe and to determine the accuracy of sound intensity measurements by comparing with sound-pressure level measurements made in the farfield of an acoustic source inside an anechoic room. The results show that there are noticeable changes in the acoustic field in the frequency range 3000–6000 Hz due to the insertion of the probe, and the scattering/diffraction effects tend to increase with increase in the angle of incidence of the sound waves, but do not affect in any significant way the accuracy of sound intensity measurements. Measurement accuracies of the probe are within 1 dB up to 5000 Hz, which includes other sources of errors associated with signal processing such as errors inherent with FFT techniques.

1:50

**QQ2. Use of sound intensity for determination of air-moving device noise emission.** A. C. Balant and George C. Maling, Jr. (Data Systems Division, International Business Machines Corporation, Poughkeepsie, NY 12602)

American National Standards on sound power determination via sound intensity and on measurement of noise from air-moving devices are at an advanced stage of development. The latter proposed standard requires a special test facility such as a reverberation room or a hemi-anechoic room, and does not permit the use of sound intensity for determination of noise emission. In this paper, the results of sound power determinations on air-moving devices in a reverberation room, a hemi-anechoic room, and in an ordinary laboratory environment are presented. The data in the ordinary room were obtained using sound intensity methods. The accuracy of the data in all three environments is discussed. Many of the "field indicators" which are recommended in the proposed sound intensity standard have been calculated in order to determine acceptable ranges for these indicators. The results indicate that determination of air-moving device sound power in ordinary rooms is adequate for many engineering purposes. Use of sound intensity for determination of air-moving device sound power should increase the amount of data available on these sources.

2:05

**QQ3. Investigation into the use of the filtered-x LMS algorithm for noise reduction in a duct.** Peter L. Schuck (Physics Division, National Research Council, Montreal Road, Ottawa, Ontario K1A 0R6, Canada)

Burgess [J. Acoust. Soc. Am. 70, 715–726 (1981)] simulated an active noise reduction system in a duct. This system utilized a version of the filtered-x LMS algorithm to obtain noise reduction over a wide bandwidth. A similar system has been implemented using the TMS-32020 signal processing chip on a PC expansion board. In the filtered-x LMS algorithm, the reference signal must be filtered by the transfer function between the transversal filter output and the error signal input in order to maintain stability of the LMS weight update. In our system, this transfer function is calculated from the measured response to a maximum length sequence generated by the noise reduction system itself. Performance results are given for this system in a test setup. Some issues regarding practical implementation of this algorithm in fixed point arithmetic will also be discussed.

2:20

**QQ4. Accurate estimation of amplitude, frequency, and phase of harmonic signal components using an FFT.** John C. Burgess (Department of Mechanical Engineering, University of Hawaii, Honolulu, HI 98622)

The signal analyzed using an FFT is seldom identical to the true signal. This can result in inaccurate estimates of amplitude, frequency, and phase of harmonic signals. Using an extension of a method described earlier [J. C. Burgess, "On digital spectral analysis of periodic signals," J. Acoust. Soc. Am. 58, 556–567 (1975)], it is possible to estimate the true amplitude, frequency, and phase of a harmonic signal with good accuracy. For example, acoustic intensity measurements may require determining the phase difference between two harmonic signals of the same amplitude and frequency. Using the method described, this phase difference often can be estimated with an error less than 0.01%. The method requires use of an optimum data window that can be expressed as a Fourier series having only a few nonzero coefficients.

2:35

**QQ5. The measurement of sound by the simultaneous use of several time constants.** Svein Arne Nordby, Steiner Bohn (Norwegian Electronics, P. O. Box 24, N-3408, Tranby, Norway), and Richard J. Peppin (Scantek, Inc., 51 Monroe Street, Suite 1606, Rockville, MD 20850)

Nonstationary or transient sounds measured by a sound level meter will produce different values of sound level, depending on the time constant, or lack thereof, used. The technique of determining exposure, the so-called "doubling rate" will also result in significant differences in readings. Most investigations of the effects of the different time constants and averaging rates on nonstationary signals required several sound level meters or a predefined signal that was repeated for each different measurement method so that the differences could be compared. This paper presents the results of tests using a sound level meter that can simultaneously measure with various time constants and with several "averaging" techniques allowing the operator to measure  $L_{max}$  and  $L_{min}$  using "fast," "slow," "impulse" time constants and "peak" detection using a doubling rate of 3, 4, 5, 6, and impulse, all on the same arbitrary source.

2:50

**QQ6. Computational studies of the diffraction integral occurring in the MAE theory of sound propagation over hills and valleys.** James A. Kearns, Ji-xun Zhou, Yves H. Berthelot, and Allan D. Pierce (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332)

An important class of problems pertaining to outdoor sound propagation is that of the diffraction that occurs when the ground is neither perfectly flat nor perfectly rigid. Such problems are encountered in the study of long-range propagation of sound over hills and valleys. It has been shown previously [J. Acoust. Soc. Am. Suppl. 1 79, S30-S31 (1986)] that the theory of matched asymptotic expansions allows one to express the diffracted field in terms of a complex integral involving Airy functions of complex argument. In some limiting cases, the diffraction integral reduces to some computationally very efficient forms: an equation based on geometrical acoustics in the illuminated region, a creeping wave series in the shadow zone, and a knife-edge Fresnel diffraction integral far behind the ridge. In the present paper, the transition between these different regimes is investigated numerically by computing the general integral, and particular attention is given to the matching with the creeping waves series solution in the penumbra region. Computational results are compared with data obtained in laboratory scaled experiments. [Work supported by NASA Langley Research Center.]

3:05

**QQ7. Sound attenuation in air over water on Puget Sound during temperature inversion and noninversion conditions.** Jan H. Hauge (Towne, Richards & Chaudiere, Inc., 105 N. E. 56th Street, Seattle, WA 98105)

Excess attenuation of sound over a 7500-ft distance was measured on 6 days to determine effects of temperature inversions and wind direction. The tests were conducted using one-third octaves of pink noise at center frequencies of 250, 500, and 1000 Hz. During temperature inversions, excess attenuation downwind was found to be significantly reduced and negative. Results of the tests were used to predict sound levels that would be received at island residences and a bird sanctuary under a proposed industrial use of the mainland shoreline.

3:20

**QQ8. Model scale testing of outdoor noise propagation from a power plant.** Robert A. Putnam (Gilbert Services, Inc., P. O. Box 1498, Reading, PA 19603)

A proposed cogenerating power plant of 70-MW capacity is to be located in an urban setting and will be subject to stringent environmental noise emission criteria. A four-cell forced draft cooling tower is included in the plant design. Computer modeling of cooling tower noise propagation to affected neighboring residences involved significant uncertainties due to the complex shapes of power plant building elements and existing surrounding buildings. Without a finished facility in which to perform full-scale field tests, the next best thing is a scale model test. This paper will discuss the specifics of the 1/20th scale test setup, materials, selection, instrumentation, test execution, some problems and some lessons learned. The results of the test will be given in terms of the differences in octave band sound-pressure levels between an unobstructed free-field value and the modeled *in situ* value, at several receptor positions. The test enabled the specification of more precise sound attenuating elements for the cooling tower fans and inlets in order to comply with the governing regulations.

THURSDAY AFTERNOON, 19 MAY 1988

EAST BALLROOM B, 1:30 TO 5:00 P.M.

### Session RR. Physical Acoustics VII: Scattering

Kevin L. Williams, Chairman

Code 4120, Naval Coastal Systems Center, Panama City, Florida 34207-5000

#### Contributed Papers

1:30

**RR1. A variational principle for the scattered wave.** D. E. Freund and R. A. Farrell (The Johns Hopkins University Applied Physics Laboratory, Johns Hopkins Road, Laurel, MD 20707)

Schwinger-type variational principles are presented for the scattered wave in the case of scalar wave scattering for an arbitrary incident field from an object of arbitrary shape with either homogeneous Dirichlet or homogeneous Neumann boundary conditions. Designating the distance from the scatterer to the observer by  $r$ , then the results are variationally invariant for all values of  $r$  ranging from the surface of the scatterer to the farfield. Explicit results are presented for the case when the scatterer is a sphere obeying homogeneous Dirichlet boundary conditions. Special attention is given to the selection of the trial fields that produce accurate results over a broad frequency range. [Work supported by the U.S. Navy.]

1:45

**RR2. An analysis of partial wave scattering coefficients and the interpretation of resonances and background in scattering from elastic targets.** M. F. Werby (NORDA Numerical Modeling Division, NSTL, MS 39529) and H. Überall (Physics Department, Catholic University of America, Washington, DC 20064)

Bistatic angular distributions and backscattered form functions can be represented as a partial wave series where the angular contribution can be written in terms of spherical harmonics or some other suitable angular function. It is possible to represent elastic, rigid, and solid targets in such a representation. The appropriate background (in partial wave space) can be subtracted from some elastic target and a resonance can be isolated. This is usually done by assuming a rigid or a soft background for elastic solids or thin shells and leads to a residual response that, under appropriate conditions, can manifest a resonance. Often, however, neither a rigid

nor a soft background is suitable. This can be established by examining the residual partial wave coefficients, which, in the event of a poor choice of background, will lead to a mix of several partial waves at resonance. On the other hand, for a good background choice, a predictable mix of partial wave coefficients will be observed. When a suitable background has been chosen, then the partial wave analysis allows one to characterize the resonance.

2:00

**RR3. An analysis of the effects of fluid loading on the vibrational modes of a submerged spherical shell.** Gary S. Sammelmann and Roger H. Hackman (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407-5000)

Previous analyses of the acoustic scattering from submerged, spherical elastic shells have relied on the subtraction of a "suitably defined" background contribution to isolate scattering resonances of the shell. Unfortunately, there is no universal agreement as to what constitutes a suitable background. In the present work, the acoustic scattering amplitude is analytically continued into the complex- $k$  plane and the poles of this amplitude are directly located using a form of Cauchy's theorem. This procedure alleviates any difficulties associated with a background choice. Dispersion curves are generated for shells with thicknesses ranging from 2% to 50% of the outer radius of the shell. This work verifies the bifurcation of the first symmetric mode of a spherical shell reported earlier [L. H. Green, R. H. Hackman, D. H. Trivett, and L. Flax, *J. Acoust. Soc. Am. Suppl. 1* 76, S65 (1984)], although the details of the bifurcation differ considerably. Our results call into question the whole concept of the separation of poles into background and elastic categories.

2:15

**RR4. GTD for backscattering from elastic objects in water: Phase of the coupling coefficient and a simplified synthesis of the form function.** P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164) and K. L. Williams (Naval Coastal Systems Center, Panama City, FL 32407)

The geometrical theory of diffraction (GTD) was recently extended to describe surface elastic wave (SEW) contributions to scattering from fluid-loaded spheres and cylinders at high frequencies [P. L. Marston, *J. Acoust. Soc. Am.* 83, 25-37 (1988)]. The coupling of an SEW with the scattered field was described by a complex coefficient  $G$ . That analysis gave simple physical approximations:  $|G| \approx 8\pi\beta c/c_{sw}$  for spheres and  $|G| \approx 8\pi\beta/(\pi ka)^{1/2}$  for cylinders. The SEW angular attenuation coefficient (due to radiation damping) and phase velocity are  $\beta$  and  $c_{sw}$ , while  $c$  denotes the sound speed in the fluid,  $k = \omega/c$ , and  $a$  is the object's radius. The present work gives approximations for the phase of  $G$  derived in the limit  $\beta \ll 1$  by assuming that SEW contributions match those of resonance scattering theory based on a rigid background. The resulting phase of  $G$  is  $\approx 0$  for solid spheres and  $\approx \pi/4$  for solid cylinders. The resulting approximation for spheres  $G \approx 8\pi\beta c/c_{sw}$  yields a synthesis of the exact form function similar in quality to previous ones [K. L. Williams and P. L. Marston, *J. Acoust. Soc. Am.* 79, 1702-1708 (1986)] based on direct numerical evaluations of  $G$ . The results suggest that, given  $\beta$  and  $c_{sw} > c$ , the SEW contribution may be accurately approximated when  $ka \gg 1$ . The new synthesis used a simple approximation for the specular contribution. [P. L. Marston was supported by ONR.]

2:30

**RR5. Utilizing the form functions for determining the physical parameters of elastic solid spheres.** N. Yen and Louis R. Dragonette (Physical Acoustics Branch, Naval Research Laboratory, Washington, DC 20375-5000)

An approach is described to determine the density and the longitudinal and the shear velocities of elastic solid spheres based on their farfield

response (the form function) in an insonified acoustic field. The processing scheme uses the information extracted from the characteristics of the form function within a finite frequency range. In particular, it is found that the rate of phase change with respect to frequency in the form function reveals three distinct types of reradiated modes, which are associated with their underlying physical causes. Their separated physical parameters can thus be determined accordingly. Examples with solid spheres constructed from different materials are used to illustrate the practical utilization of this study.

2:45

**RR6. Identification of whispering gallery type waves on a finite cylindrical shell using MIIR.** S. K. Numrich (Code 5133, U.S. Naval Research Laboratory, Washington, DC 20375-5000), G. Maze, J. L. Izbicki, and J. Ripoché (Laboratoire d'Electronique et d'Automatique, Université du Havre, Place Robert Schuman, 76 610 Le Havre, France)

Experiments were conducted at Laboratoire d'Electronique et d'Automatique (ULTRASONS) to confirm the presence of whispering gallery type waves on a finite cylindrical shell bounded by hemispherical endcaps. The method of isolation and identification of resonances (MIIR) developed by the research team at ULTRASONS was used to obtain the bistatic mode patterns that conclusively identified the modes observed monostatically by researchers at the Naval Research Laboratory. The radiation patterns for the several low-order whispering gallery modes were identifiable when the cylindrical shell was insonified normal to its major axis. Unequivocal identification of higher-order modes was inhibited by the presence of waves traveling along other circumferential paths.

3:00

**RR7. Broadside and end-on resonance scattering from submerged high-aspect ratio targets for several elastic materials.** H. Überall (Physics Department, Catholic University of America, Washington, DC 20064) and M. F. Werby (NORDA Numerical Modeling Division, NSTL, MS 39529)

Broadside and end-on scattering from solid elastic spheroids composed of aluminum, steel, and WC is examined. The spheroids range in aspect ratio from 4 to 10 with a  $kL/2$  range out to 26. A comparison of the form functions for the different materials shows that the ratios of the  $kL/2$  values at resonances are equal to the ratio of the Rayleigh phase velocities of the respective materials and that the end-on and broadside resonance locations are in accordance with the phase matching method of Überall *et al.*, which is based on the leaky Rayleigh type wave interpretation of resonances. Further, it is shown that the resonance locations are only sensitive to the shear speed and that it is possible to generate resonances that occur end-on at all angles including broadside, which is only consistent with the Rayleigh wave interpretation of resonances.

3:15

**RR8. Acoustic scattering from large-aspect ratio, axisymmetric shells: II.** Roger H. Hackman and Douglas G. Todoroff (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407-5000)

Previously [R. H. Hackman and D. G. Todoroff, *J. Acoust. Soc. Am. Suppl. 1* 82, S49 (1987)], a comparison was presented of the predictions of the spheroidal-coordinate based  $T$  matrix with preliminary measurements of the acoustic scattering from a copper, prolate spheroidal shell at axial incidence. This target had an aspect ratio of  $L/D = 7$  and a constant thickness of 5% of the semiminor axis. Here, more detailed measurements are presented and both the calculations and the measurements to general incidence are extended. The emphasis of this work is the physical analysis of the elastic excitations of the scatterer. The predictions of the spheroidal-coordinate based  $T$  matrix are also compared with previously pub-

3:30

**RR9. Multiple scattering analysis for a target in an oceanic waveguide.** Roger H. Hackman and Gary S. Sammelmann (Physical Acoustics Branch, Naval Coastal Systems Center, Panama City, FL 32407-5000)

A multiple scattering analysis is presented for a target in a range-independent oceanic waveguide. The multiple scattering series is explicitly summed and a solution is obtained in closed form. The solution agrees with that presented earlier [R. H. Hackman and G. S. Sammelmann, *J. Acoust. Soc. Am.* **80**, 1447-1458 (1986)] for all cases that were checked. The method provides an attractive alternative to the previous formalism in two regards. First, this approach provides insight into the algebraic structure of the solution. The individual terms in the final solution may be simply interpreted in terms of single scattering processes. And, second, there is less algebra involved when relatively simple waveguides are considered. Numerical examples are given that illustrate the importance of contributions involving rescattering among the target and the waveguide boundaries.

3:45

**RR10. Acoustic resonance spectra of a finite cylindrical shell** G. Maze, F. Lecroq, J. L. Izbicki, and J. Ripoche (Laboratoire d'Electronique et d'Automatique, Université Le Havre, Place Robert Schuman, 76610 Le Havre, France)

Previous studies have explained acoustic scattering from an infinite cylinder or a sphere. The scattering depends strongly on resonances that are related to the propagation of circumferential waves when the acoustic axis is perpendicular to the cylinder axis or the propagation of guided waves when the insonification is oblique. The method of isolation and identification of resonances (MIIR) developed in our laboratory is used to study acoustic scattering from a finite cylindrical shell bounded by two plane disks. The acoustic beam diameter is larger than the target dimensions. When the acoustic beam axis is perpendicular to the target axis, the resonances observed on the resonance spectrum are identical to those observed with an infinite cylindrical shell. When the acoustic beam axis is not perpendicular to the target axis, the resonances are more numerous and, for certain incidence angles, next resonances have the same mode.

4:00

**RR11. Bounded beam transmission across water/sand interface. Part I—Experiment.** L. J. Satkowiak and K. L. Williams (Naval Coastal Systems Center, Code 4120, Panama City, FL 32407-5000)

A series of measurements was made to investigate the effect of the water/sediment interface on a propagating acoustic signal. Under nearly laboratory conditions, data were collected mapping out the sound-pressure field in a homogeneous sand for a variety of nonlinear and linear sources. The data were collected using the following sources: a truncated parametric source, a saturation limited parametric source with effective length less than the sender-interface distances, and a linear source, all operating at a frequency of 20 kHz. The data were taken at grazing angles that ranged from well-above to well-below the critical angle for the sediment. Based on the data collected from the in-sediment pressure field, contours were obtained detailing the magnitude and direction of the pressure wave in the sediment. These pressure field contours, as well as experimental procedures and analysis techniques, will be presented. A comparison with theoretical predictions will be made in the following paper.

**RR12. Bounded beam transmission across water/sand interface. Part II—Experiment versus theory.** K. L. Williams and L. J. Satkowiak (Naval Coastal Systems Center, Code 4120, Panama City, FL 32407-5000)

The experiments described in the previous presentation allow the generation of contour plots of the pressure field within the sediment for different types of sending transducers at several grazing angles (defined here as the angle between the interface and central axis of the sender). These contour plots are compared with theoretical predictions calculated using the SAFARI model [H. Schmidt and F. B. Jensen, *J. Acoust. Soc. Am.* **77**, 813-825 (1985)]. Results indicate that the shear degree of freedom in the sediment is unimportant under the conditions of the experiment. Agreement between SAFARI and experimental pressure field contours is good for all senders used in the experiment. This suggests that nonlinear effects due to parametric array truncation were not readily apparent in the contour plots. Based on the experimentally determined in-sediment direction of propagation, the presence or absence of beam displacement for each type of acoustic source will be discussed.

4:30

**RR13. An iterative solution to acoustic scattering by rigid bodies.** Luise Schuetz, Joseph J. Shirron, and Ralph Kleinman (Naval Research Laboratory, Washington, DC 20375)

The scattering problem is formulated as an integral equation over the scattering surface. A solution to the equation is given in a convergent Neumann series where the convergence depends on a parameter  $\alpha$ . The boundary element method is used to obtain an approximation to the solution, and the resulting matrix equation is solved by an iterative technique. Results are given for several different geometries and frequencies. A comparison of solution time and accuracy to direct methods such as Gaussian elimination is presented, showing the iteration to be more economical in most cases of practical interest. Finally, there is a short discussion on determining the iteration parameter  $\alpha$  and its effects on the rate of convergence.

4:45

**RR14. Computations of rigid body scattering by long, slender finite cylinders using the  $T$ -matrix method.** Angie Sarkissian (Sachs/Freeman Associates, 1401 McCormick Drive, Landover, MD 20785)

Waterman's  $T$ -matrix algorithm requires the evaluation of matrix elements  $Q_{lk}$  whose imaginary parts are difficult to evaluate for  $l < k$  for long, slender scatterers, at high frequencies, because they involve surface integrals of large oscillatory functions. Machine precision required for numerical integration of these matrix elements increases as the frequency is increased. Therefore, the frequency range over which the method can be applied is usually limited by the precision of the computer used. Certain shapes, such as ellipsoids, have symmetric  $Q$  matrices; since numerical integration of the difficult side of the matrix can be avoided for these shapes, the frequency range over which the method can be applied seems to be limited by machine memory alone. A different approach for the evaluation of the imaginary part of  $Q$  will be shown for finite cylinders where the antisymmetric part of the matrix is written in a form involving surface integrals that are more easily computed. By avoiding integration of large oscillatory functions, this method overcomes the difficulties associated with long slender cylinders and enables high-frequency computations. [Work supported by Naval Research Laboratory, Washington, DC.]

**Session SS. Physiological Acoustics III: Structure and Function: The Ear Canal to the Auditory Nerve**

Elizabeth C. Oesterle, Chairman

*Department of Otolaryngology, University of Washington, Seattle, Washington 98195***Contributed Papers**

1:30

**SS1. Coupling between ear canal and eardrum in model systems.** Michael R. Stinson (Division of Physics, National Research Council, Ottawa K1A 0R6, Canada) and Shyam M. Khanna (Department of Otolaryngology, Columbia University, New York, NY 10032)

This paper examines a theoretical model of sound propagation in the ear canal that accounts for the *distributed* acoustical load presented by the eardrum. This model is an extension of our previous one-dimensional formulation [S. M. Khanna and M. R. Stinson, *J. Acoust. Soc. Am.* **77**, 577-589 (1985)], which was limited to rigid-walled canals. The spatially varying motion of the eardrum enters as a driving term in the equation for the sound field. In principle, this present model can be combined with detailed, dynamical models of the tympanic membrane. Experiments using model canals are underway to test the theory. In these, the sound field is determined with the eardrum replaced by either a mechanically driven piston or a passive, locally reacting load. Preliminary results indicate that the theoretical formulation can describe the sound field in the cat ear canal up to at least 25 kHz and in the human ear canal up to at least 15 kHz.

1:45

**SS2. Determination of the reflection coefficient and the cross-sectional area function from high-frequency pressure measurements in the ear canal.** R. D. Rabbitt (Department of Mechanical Engineering, Washington University, St. Louis, MO 63130)

At moderate to high frequencies, incoming and outgoing pressure waves combine to form standing waves in the ear canal. The exact form of the pressure distribution depends on the geometry of the ear canal and on the magnitude, and phase, of the reflected wave relative to the incoming wave. An asymptotic theory describing this interaction is combined with pressure measurements in the ear canal in order to determine the reflection coefficient at the eardrum, the cross-sectional area function of the ear canal, and the relative phase of the reflected wave. The theory is based on a high-frequency multiscale solution of the one-dimensional horn equation and is shown to agree well with experimental measurements of standing wave patterns and with the distribution of phase in the ear canal. Since the absorption of energy at the eardrum is determined, this method is particularly well suited to determine the input to the ear at high frequencies. [Work supported, in part, by the Whitaker Foundation.]

2:00

**SS3. Measured frequency responses of the peripheral auditory organs in goldfish.** Mardi C. Hastings and Peter H. Rogers (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405)

The frequency responses of the primary peripheral auditory organs in eight goldfish (*Carassius auratus*) were measured using a recently developed noninvasive vibration detection system based on ultrasound and having a spatial resolution of 0.28 mm [M. Cox and P. H. Rogers, *J. Vib. Acoust. Stress Reliability Design* **109**, 55-59 (1987)]. The fish averaged

4.5 cm in standard body length and 3 g in weight. They were freely suspended underwater and anesthetized during the measurement process. Most of the responses were recorded *in vivo*; however, some of the fish did not recover from the anesthesia and may have died during the procedure. The results are the first to verify the existence of a high-pass filtering mechanism via the Weberian ossicles, as previously postulated in the literature. [Work supported by ONR.]

2:15

**SS4. Model calculations of self- and external-tone interactions of otoacoustic emissions.** Savithri Sivaramakrishnan, Glenis R. Long (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907), and Arnold Tubis (Department of Physics, Purdue University, West Lafayette, IN 47907)

Several different forms of nonlinear, active, cochlear models are used to simulate some of the effects of interactions among spontaneous otoacoustic emissions in the same ear [E. M. Burns, E. A. Strickland, A. Tubis, and K. Jones, *Hear. Res.* **16**, 271-278 (1984)] and interactions of emissions with external tones. The former include mutual suppression, and intermodulation distortion products, while the latter include the suppression, frequency shifting, and frequency locking of spontaneous emissions, and the latencies of click and tone-burst evoked emissions. [Work supported by NIH Grant No. NS22095.]

2:30

**SS5. Distribution of two major organ of Corti proteins in different component cell types.** Peter Kraus, Jogy Varghese, Isolde Thalmann, Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110), and H.-P. Zenner (Department of Otolaryngology, University of Würzburg, D-8700 Würzburg, Federal Republic of Germany)

Two highly acidic, low molecular weight proteins, present in very high concentration in the organ of Corti (OC) and termed OCP-I and OCP-II [I. Thalmann *et al.*, *Arch. Otorhinolaryngol.* **226**, 123-128 (1980)] have been previously described. The proteins do not contain significant amounts of carbohydrate and are soluble in both high- and low-ionic strength buffer. The distribution of these unidentified proteins in different cell types of the OC has now been determined and has been compared with the distribution of actin, tubulin, and calmodulin. Freeze-dried and freshly dissected substructures of the OC of chinchilla and guinea pig were used, respectively. Actin was found to be particularly abundant in the inner and outer pillar cells. In the Hensen and Deiters cells, OCP-I and OCP-II predominate and calmodulin is present only in very small quantities. In contrast, the outer hair cells exhibit a strong accumulation of calmodulin, with relatively low levels of OCP-I and OCP-II, actin and tubulin. In the neuroepithelium of the sacculus OCP-I and OCP-II are present at about 40 times lower concentrations than in the OC.

**SS6. Behavior of two major organ of Corti proteins during development and maturation of the cochlea.** Isolde Thalmann, Peter Kraus, Jogy Varghese, Thomas H. Comegys, Ruediger Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110), and H.-P. Zenner (Department of Otolaryngology, University of Würzburg, D-8700 Würzburg, Federal Republic of Germany)

Two highly acidic, low molecular weight proteins, present in very high concentration in the organ of Corti (OC) and termed OCP-I and OCP-II [I. Thalmann *et al.*, *Arch. Otorhinolaryngol.* **226**, 123–128 (1980)] have been previously described. The behavior of these proteins during the post-

natal development in the rat cochlea has now been determined. On the 6th day post partum (pp), only trace amounts of the substances are present. The concentration of OCP-I and OCP-II increases most rapidly between days 12 and 14 pp, and the proteins approach adult levels by day 18 pp. The sharp increase of OCP-I and OCP-II occurs at approximately the same time as the sudden increase of the endolymphatic potential on day 13 pp [S. K. Boshier and R. L. Warren, *J. Physiol.* **212**, 739–761 (1971)]. It is of interest that a major basilar papilla-specific protein in the chick increases in step with the onset of auditory function during embryonic development [J. C. Oberholtzer *et al.*, *Hear. Res.* **23**, 161–168 (1986)]. This protein is similar but *not* identical in its electrophoretic pattern to OCP-II. The behavior of OCP-I and OCP-II will be compared with the pattern of other important proteins such as actin, tubulin, and calmodulin.

3:00–3:15  
Break

3:15

**SS7. Perturbation of cochlear microphonics by an intracellular electrode in the organ of Corti.** J. J. Zwislocki and R. L. Smith (Institute for Sensory Research, Syracuse University, Syracuse, NY 13244-5290)

Recording the receptor potentials of cochlear inner and outer hair cells *in situ* has become an essential part of cochlear research. To reach the cells, the recording electrode has to traverse parts of the organ of Corti. Such intrusion may affect the cochlear micromechanics. This possibility was tested by recording scala-media CM in the second cochlear turn in Mongolian gerbils. A small hole was drilled in the cochlear bony capsule above the basilar membrane, and an intracellular electrode was made to traverse Hensen's cells all the way to the OHCs. A second electrode placed in the endolymph recorded CM amplitude and phase as a function of sound frequency. Our results indicate that an intracellular electrode lodged in the organ of Corti can produce substantial changes in CM tuning. This means that the cochlear micromechanics was perturbed. [Work supported by Javits Grant NS 03950.]

3:30

**SS8. Indirect evidence for the absence of electrotonic coupling between hair and support cells in the mammalian cochlea.** Elizabeth Oesterle and Peter Dallos (Department of Otolaryngology, RL-30, University of Washington, Seattle, WA 98195 and Auditory Physiology Laboratory and Department of Neurobiology and Physiology, Northwestern University, Evanston, IL 60201)

Gap-junctional complexes link goldfish and lizard hair cells with adjacent supporting cells [Hama, *J. Neurocytol.* **9**, 845–860 (1980); Nadol, Mulroy, Goodenough, and Weiss, *Am. J. Anat.* **147**, 281–302 (1976)]. Lizard hair cells and support cells appear to be electrically coupled [Baden-Kristensen and Weiss, *Hear. Res.* **8**, 295–315 (1982)]. In mammals, anatomical evidence suggests the absence of gap-junctional complexes between hair cells and support-cell neighbors, but suggestive membrane specializations have been observed [Gulley and Reese, *J. Neurocytol.* **5**, 479–507 (1976); Nadol, *Ann. Otol.* **87**, 70–80 (1978)]. Horseradish peroxidase marking and intracellular recording techniques were utilized to determine the origin(s) of sound-induced potential changes in supporting cells of the organ of Corti. Recordings were made in the low-frequency region of the adult guinea-pig cochlea, an *in vivo* preparation. Response characteristics of support cells to tone bursts at various sound levels and frequencies were compared with hair-cell responses (IHC and OHC) and potentials recorded in organ fluid spaces. Comparisons of the harmonic content of the responses, magnitude, and phase of the fundamental component, and the magnitude of dc-response components suggest a lack of electrotonic coupling between mammalian hair cells and support cells. [Work supported by NIH Grant NS-08635.]

3:45

**SS9. Neurotransmitter and peptide neuromodulator candidates in hair-cell and nerve fractions of the saccular macula of the rainbow trout.** Marian J. Drescher and Dennis G. Drescher (Laboratory of Biotechnology, Wayne State Medical School, 540 E. Canfield Avenue, Detroit, MI 48201)

An epithelial sheet of 30 000–40 000 hair cells, detached from the basal lamina of the trout saccular macula, has been analyzed for primary-amine and N-acetylated compounds by cation-exchange HPLC. This hair-cell fraction (HC), greatly reduced in supporting cells and neural elements, was simpler in primary-amine composition than the saccular nerve fraction (SN), which included afferent and efferent neurons and Schwann cells. The HC contained more phosphoserine and phosphoethanolamine, or co-eluting components, per microgram protein than SN [cf. M. J. Drescher and D. G. Drescher, *Comp. Biochem. Physiol.* **86A**, 553–558 (1987)], and ten times more beta-alanine. Of the excitatory amino acids in HC, glutamate was present in highest concentration, and was significantly elevated ( $p < 0.02$ ), 2.6-fold, above levels determined for SN, consistent with published efflux and release studies [M. J. Drescher, D. G. Drescher, and J. S. Hatfield, *Brain Res.* **417**, 39–50 (1987)]. Histidine-containing peptides in millimolar concentrations appeared uniquely partitioned: Carnosine and homocarnosine, in a ratio of 10:1, respectively, were associated with SN, while a third histidine-containing component, tentatively identified as N-acetyl histidine, was associated primarily with HC. Possible involvement of histidine-containing compounds as neuromodulators will be discussed. [Work supported by NIH Grant NS 16166.]

4:00

**SS10. Quantitative light microscopic comparison of cochlear nerve morphology in Sprague-Dawley and Brown Norway rats.** Virginia Hoeffding and Martin L. Feldman (Department of Anatomy, Boston University School of Medicine, 80 E. Concord Street, Boston, MA 02118)

Cochlear nerve morphology was studied in young adult albino (Sprague-Dawley) and pigmented (Brown Norway) rats. Following perfusion of the animals with mixed aldehydes, blocks including the cochlear nerves were removed, embedded in Araldite, and sectioned in a plane transverse to the longitudinal axis of the nerve. Analysis of the material included counts of normal and degenerating fibers and of glial cell nuclei, and measurements of vascularity and of the nerve cross-sectional area. The median number of normal fibers in the Sprague-Dawley rats was 21 216; in the Brown Norway rats it was 20 186. There were no statistically significant differences between the two strains in numbers of normal fibers, degenerating myelin sheaths, or glial cell nuclei, or in the cross-sectional areas of the nerves. Vascular area density ( $V_{AD}$ ) was significantly higher

in the Sprague-Dawley rats. The median ( $V_{AD}$ ) in that strain was 0.0149, while in the Brown Norway rats the median ( $V_{AD}$ ) was 0.0099. [Work supported by NIH Grants AG 00001 and T32 NS07152.]

4:15

**SS11. Forward masking of the compound action potential: Thresholds for the detection of the N1 peak.** Evan M. Relkin and Robert L. Smith (Institute for Sensory Research and Department of Bioengineering, Syracuse University, Syracuse, NY 13244-5290)

A computerized two-interval forced-choice method was used to measure the threshold for detection of the N1 peak of the compound action potential (CAP) recorded at the round window of the chinchilla in response to a 23-ms probe tone. A 100-ms, pure-tone masker of equal frequency preceded the probe tone. Threshold shifts produced by the masker were measured as the intensity of the masker and the time delay between the masker and the probe were varied. The techniques differ from prior studies in that forward masking of the compound action potential has been quantified as shifts in detection thresholds rather than decrements in the magnitude of the N1 peak. Results were compared to analogous measures for single neurons in the auditory nerve [E. M. Relkin and D. G. Pelli, *J. Acoust. Soc. Am.* **82**, 1679-1691 (1987)] and analogous psychophysical measures of forward masking. Preliminary findings indicate that forward masking of the CAP more closely corresponds to that observed psychophysically than does the forward masking observed in the response

of a single neuron. [Work supported by the Deafness Research Foundation and NIH.]

4:30

**SS12. Auditory nerve fiber responses predict the discriminability of VOT syllables.** Donal G. Sinex and Lynn P. McDonald (Los Alamos National Laboratory, LS-1 M882, Los Alamos, NM 87545)

Responses of chinchilla auditory nerve fibers to synthesized syllables differing in VOT were obtained. A continuum heard as /da~/~/ta~/ was used. Many neurons responded with an increase in average discharge rate or with a change in synchronized response whose latency matched the nominal stimulus VOT. Response latency was treated as a random variable. For each VOT pair, a statistic analogous to  $d'$  was calculated from the distribution of the difference between two random variables. Thresholds for  $\Delta$ VOT were interpolated from psychometric functions based on this statistic. Thresholds were lowest for standard VOTs of 30-40 ms, but were highly dependent on the direction of change from the standard. These characteristics were also typical of the chinchilla's psychophysical thresholds for  $\Delta$ VOT, reported by Kuhl [*J. Acoust. Soc. Am.* **70**, 340-349 (1981)]; in addition, the quantitative agreement between neural and behavioral thresholds was usually very good. For this VOT continuum, discriminability reflects the precision of stimulus coding at the level of the auditory nerve. [Work supported by NINCDS.]

THURSDAY AFTERNOON, 19 MAY 1988

418/420 ROOM, 1:30 P.M.

### Meeting of Accredited Standards Committee S3 on Bioacoustics

L. A. Wilber, Chairman S3

422 Skokie Boulevard, Wilmette, Illinois 60091

**Standards Committee S3 on Bioacoustics.** The current status of standards under preparation will be discussed. In addition to those topics of interest including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards that might be needed over the next few years. Open discussion of Committee reports is encouraged.

The international activities in ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, IEC/TC 87 Ultrasonics, ISO/TC 108/SC4 and ISO/TC 94/SC12 will be discussed. The Chairs of the TAGs for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelnitsky), IEC/TC 87 (P. D. Edmonds), ISO/TC 108/SC4 and ISO/TC 94/SC12 (H. E. von Gierke) will report on current activities of these Technical Committees or Subcommittees.

## Session TT. Structural Acoustics and Vibration V: Characterization of Source Structure Interaction and Radiation Fields

Sabih I. Hayek, Chairman

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

### Contributed Papers

2:00

**TT1. The Poynting vector and source identification for thin plates and shells.** Earl G. Williams, Phillip B. Abraham (Naval Research Laboratory, Code 5137, Washington, DC 20375-5000), and Anthony J. Romano (Sachs/Freeman Associates, Inc., 1401 McCormick Drive, Landover, MD 20784)

The Poynting vector  $\mathbf{I}$ , well known in the three-dimensional theory of elasticity, represents the three-dimensional power flow in a solid. It is shown how  $\mathbf{I}$  can be used to derive the power flow in a thin flat plate, and a thin cylindrical shell. The divergence of  $\mathbf{I}$  provides the rate of change of total energy density (the Hamiltonian density,  $\partial H/\partial t$ ) in the shell. Its time average must vanish for steady-state conditions. This leads to the conclusion that power flow in and out of the structure is given by  $\Pi = \oint (\mathbf{I}_{2D}) \cdot \hat{n} dl$ , a line integral (over the shell surface) of the in-plane, average intensity in the shell. If the shell is point driven, then  $\Pi$  is large at that point, and can be used to locate the position and strength of the mechanical source. This provides an excellent tool for mechanical source identification. These concepts will be demonstrated and verified using synthetic and real data on point driven cylindrical shells in water. In the synthetic data, the full displacement vector  $(u, v, w)$  is specified throughout the shell. In the real data, measurements of the radial component ( $w$ ) of shell motion are used. In the latter case (only  $w$  known), it will be shown how it is possible, however, to get a good estimate of  $\Pi$ , and thus locate the mechanical source.

2:15

**TT2. Uncertainty bounds for energy of imperfectly characterized vibration fields.** Jean-Louis Guyader<sup>1</sup> (Lab. Vibrations-Acoustique Institut National des Sciences Appliquées de Lyon, 69621 Villeurbanne, France)

Uncertainty bounds of vibration energy for viscoelastic continuous media are given. They are related to a residual functional measuring the error of characterization associated with an approximate solution and a factor that can amplify or attenuate the uncertainty bounds, depending on the driving frequency. (That factor considerably amplifies the uncertainty when the structure is excited near resonances.) Assuming a statistical distribution of the error of characterization of the vibration field, over the modes of the structure, allows a considerable reduction of the energy uncertainty bounds. The case of Gaussian distribution is presented and analyzed for various mean values and standard deviations. The bounds can be used as a criterion of convergence of modal expansion. The case of the acoustical response of a room is presented, and the influence of the size and type, of the sets of modes taken into account in calculations, is analyzed.<sup>2)</sup> Actually on sabbatical leave at Groupe d'Acoustique de l'Université de Sherbrooke, Sherbrooke, Québec J1K 2R1, Canada.

2:30

**TT3. Response of a van der Pol oscillator to modulated sinusoidal input.** Huw G. Davies and Dennis Nandlall (Department of Mechanical Engineering, University of New Brunswick, Fredericton, New Brunswick E3B 5A3, Canada)

The response of a van der Pol oscillator with nonlinear damping and an external periodically modulated sinusoidal input is analyzed by using the method of multiple time scales. The modulation appears to provide extended stability. At resonance, the response envelope is highly nonlinear. Away from resonance the modulation causes a periodic switching on and off of a natural frequency limit cycle superimposed on the excitation frequency response. The natural frequency component arises through a parametric excitation. Extensive numerical simulations confirm the analytic results. Stability studies and the application to random modulations are also discussed. [Work supported by NSERC Canada.]

2:45

**TT4. Effects of intensity modulations on the power spectra of random processes.** William D. Mark (Physical Sciences Division, BBN Laboratories Inc., 10 Moulton Street, Cambridge, MA 02238)

An intensity-modulated random process  $\{x(t)\}$  is defined as the product of a deterministic modulating function  $\sigma(t)$  or modulating process  $\{\sigma(t)\}$  and a stationary-modulated process  $\{z(t)\}$  that is statistically independent of  $\{\sigma(t)\}$ . General expressions for the instantaneous power spectra of intensity-modulated processes are presented for various classes of modulating functions and processes. A series expansion of the instantaneous power spectrum of intensity-modulated processes is presented which has for its first term a well-known locally stationary spectrum approximation. This expansion is especially useful when the fluctuation scales  $T_o$  of the modulating functions are large in comparison with the fluctuation scales  $T_z$  of the modulated processes. The expansion can be interpreted as an asymptotic series in the parameter  $T_o/T_z$ . For given scales  $T_o$  and  $T_z$ , it is shown that modulating processes  $\{\sigma(t)\}$  possessing no first derivative have a substantially larger effect on the power spectrum of modulated processes  $\{z(t)\}$  than modulating processes possessing a first derivative. Examples illustrating various aspects of the theory are provided.

3:00

**TT5. The design, construction, and testing of a miniature shaker and force gauge.** H. Davis, B. Watters, and T. Graham (BBN Laboratories Inc., Union Station, New London, CT 06320)

BBN has designed, fabricated, and tested a miniature combined shaker and force gauge for mechanical impedance testing of light struc-



ture (metal plating of the order of 0.030 in., for example). The design is basically a Wilcoxon miniature accelerometer to be driven electrically through the signal cables, modified with a heavier reaction mass and a substitution of PZT 4 for the original PZT 5A. The shaker is attached to a stinger wire that supports the gravity load of the shaker while isolating the test object from the rotary impedance of the shaker itself. The stinger wire is attached to a specially fabricated force gauge that is glued to the test object to minimize the mass below the force gauge. The force gauge is composed of two PZT 5A crystals mounted back to back. The shaker design will be described and results presented showing that the shaker can apply up to 150 mN force with a sensitivity of 1 mN/V. The force gauge sensitivity is 350 mV/N and the frequency range is 2.6–6.6 kHz. By starting out with an accelerometer design, BBN was able to produce a working miniature shaker in a minimum amount of time.

3:15

**TT6. Random response of a Duffing oscillator: The probability density function.** Qiang Liu and Huw G. Davies (Department of Mechanical Engineering, University of New Brunswick, Fredericton, New Brunswick E3B 5A3, Canada)

A Duffing oscillator excited by narrow-band random noise can exhibit multivalued behavior, with random jumps between two pseudostable states. In this paper, an approximate form of the probability density function of the envelope of the response is obtained. State equations for the response envelope with the random excitation envelope as input are obtained by multiple time-scale analysis. The joint pdf of response envelope and input envelope can be found approximately for certain ranges of the system parameters. The response envelope pdf shows two maxima in the cases where the response is multiple valued. Numerical simulations confirm the results. [Work supported by NSERC Canada.]

3:30

**TT7. Recommended seismic zonation for Hawaii.** W. M. Adams, A. S. Furumoto, and E. Herrero-Bervera (HIG-410 A, University of Hawaii, Honolulu, HI 96822)

The state of Hawaii has both volcanic and tectonic earthquakes: The volcanic earthquakes are confined to the Big Island of Hawaii with the tectonic earthquakes occurring at all the major islands, from Hawaii to Kauai. The historical seismicity is relatively high, as might be expected from the presence of active volcanoes and the transection of the Hawaiian archipelago by the Molokai Fracture Zone. For the past 120 years, the extreme seismic intensity varies island to island: Kauai, IV; Oahu, VII; Molokai, VIII; Maui, VIII; Lanai, IX; and, for Hawaii, XII [H. D. Wood, *J. Bull. Seismol. Soc. Am.* 4, 169–203 (1914)]. This information suggests that an appropriate seismic zonation for building design is: zone 1 for Kauai and Niihau; zone 2 for Oahu; zone 3 for the islands of Maui, Lanai, and Molokai; and zone 4 for Hawaii. By estimating the loss distribution suitable for a particular location, one can derive risk functions useful for setting insurance levels [C. A. Williams and R. M. Heins, *Risk Management and Insurance* (McGraw-Hill, New York, 1985)].

3:45

**TT8. Scattering of sound from concentric cylindrical shells.** Adnan Akay (Department of Mechanical Engineering, Wayne State University, Detroit, MI 48202)

An analytical model of sound scattering from concentric, coaxial thin cylindrical shells is developed. The specific model for scattering from three concentric shells is examined in detail, and it is shown that the results can be easily extended for any number of shells. In order to see the effects of the shells and the annular fluid media between them more clearly, the case of scattering from shells of infinite length with a common axis parallel to the wave front of the incident plane wave is considered. The problem is modeled in general terms using radiation and mechanical im-

pedances. The influence of fluid loading of the shells on the resonant scattering from the multiple-shell structure is addressed. Transition from pressure boundary conditions to velocity boundary conditions is discussed and demonstrated with numerical results. Finally, effects of compliant coatings on the shells are examined. [Work supported, in part, by the National Science Foundation.]

4:00

**TT9. Application of diffraction tomography to acoustic intensity measurement.** Woon S. Gan (Acoustical Services Pte Ltd., 29 Telok Ayer Street, Singapore 0104, Republic of Singapore)

Diffraction tomography is applied to measure acoustic intensity in three-dimensional space. With Rytov's approximation, the homogeneous Helmholtz equation is obtained. The backpropagation algorithm is used to reconstruct the scattered field. Taking the special case of incident plane waves, the image kernel is expressed in convolution form, and reconstruction becomes a deconvolution process. An experimental setup is proposed. The object is a vibrating plate driven at its center. The acoustic pressure is measured from various angles. The pressure measurement acts as a diffraction tomogram. The particle velocity is reconstructed. The structural intensity (SI) is then computed using the finite difference method in terms of particle velocity. A cross-section image of the SI is obtained, which is an advantage over the nearfield acoustic holography method of J. D. Maynard, E. G. Williams, and Y. Lee [*J. Acoust. Soc. Am.* 78, 1395–1413 (1985)], which can only give the acoustic intensity as contours and three-dimensional plots. From the cross-section image, one will be able to clarify the outstanding theoretical problems of active intensity and reactive intensity.

4:15

**TT10. Diagnostic spectra catalogs—A powerful means for prediction of faults in machinery.** M. Moustafa (Faculty of Engineering, Alexandria University, Alexandria, Egypt), M. Shabara, A. El-Bahloul, and M. Fanny (Faculty of Engineering, El-Mansoura University, El-Mansoura, Egypt)

A well-designed joint will produce low levels of inherent vibration. As the joint wears, however, clearance between mating components increases and this is reflected in an increase in the vibration energy. As this vibration energy is dissipated through the mechanism links, it excites resonances and exerts extra dynamic loads on joints. It is of great interest to the maintenance engineers to predict the occurrence of excessive clearance in joints, regarding its location and magnitude. The fact that vibration signals carry much information relating to the running conditions of joints can be used to solve this problem. In this study, a mathematical model is formulated to simulate the effect of clearance in joints on the vibration spectrum of machines and mechanisms. The equations of motion of the mechanism frame and links are formulated and solved simultaneously using the Runge-Kutta method. In doing so, and with the aid of fast Fourier transform (FFT), the frequency spectra of the frame in the horizontal, vertical, and angular directions are obtained. The spectra obtained for the machine corresponding to various faults with different locations and magnitudes can be used to construct vibration charts which form the "diagnostic spectra catalog." This DSC can be used to determine the proper times for maintenance and the preparation of spare parts to avoid failure of the machines.

4:30

**TT11. Effect of backlash on vibration spectrum in gear boxes.** M. Moustafa (Faculty of Engineering, Alexandria University, Egypt), M. Shabara, Y. Abo-Mosallam, and F. Al-Saeidy (Faculty of Engineering, El-Mansoura University, Egypt)

An ideal gear box would produce no vibration at all because all energy would be channeled into the work to be done; but a well-designed gear box

will produce low levels of inherent vibration. As the gear teeth wear, however, backlash between meshing teeth increases and this is reflected in an increase in the vibration energy. As the vibration energy is dissipated through the gear box, it excites resonances and exerts extra dynamic loads on gear teeth. It is of great interest to the maintenance engineers to predict the occurrence of backlash between gear teeth. The fact that vibration signals carry much information relating to running conditions of gear boxes can be used to solve this problem. In this study, a mathematical model was formulated to simulate the effect of backlash between gear teeth on the vibration spectrum of gear boxes. A single stage helical gear

box was used to demonstrate this model. In this simulation, the transmission shafts were treated as lumped parameter systems. The elasticity of shafts, bearings, and gear teeth were also included. The grounded-chair representation was used to obtain the equations of motion that were solved using the Runge-Kutta method. The vibration spectra of the gear box system were obtained using fast Fourier transform (FFT). These spectra can be used to construct a vibration chart which, in turn, can be used to determine the proper times for maintenance of gear boxes to avoid failure of such units.

THURSDAY AFTERNOON, 19 MAY 1988

WEST BALLROOM B, 3:00 TO 5:05 P.M.

## Session UU. Underwater Acoustics VII: Signal Processing Methods

David J. Thomson, Chairman

*Defence Research Establishment Pacific, FMO Victoria, British Columbia V8W 2Y2, Canada*

Chairman's Introduction—3:00

### Contributed Papers

3:05

**UU1. Underwater active array for echo cancellation. Application to reciprocal measurements.** Gilles Chatel (Société d'Etudes et de Contrôle en Acoustique et Vibration, 6, Bd. de Brazza, 13008 Marseille, France), Thierry Rohan (Centre d'Etude et de Recherches pour la Discrétion Acoustique des Navires, Cerdan, Dcan Toulon, 83800 Toulon Naval, France), Jean-Pierre Pasqualini (Consulting Engineer for Cerdan, Dcan Toulon, 83800 Toulon Naval, France), and Alain Roure (Laboratoire de Mécanique et d'Acoustique, CNRS, 13009 Marseille, France)

Reciprocal measurement of mechanical-acoustical transfer functions is often used. If the radiating structure is immersed in shallow water, the problem of parasitic reflections arises. A new method has therefore been developed, based on the use of a reciprocal array of acoustic sources. This method allows real-time measurement of free field structure radiation on a wide frequency range. Four acoustic sources are monitored so as to create within an observation volume an acoustic field equivalent to the one generated by a monopolar source placed in the free field. The excitation signals are calculated by a stationarity method, taking into account the intrinsic characteristics of the environment. First, tests carried out in shallow water enabled us to check that the acoustic pressure field measured was the one required. Second, this method was applied to determine by reciprocity the free field mechanical-acoustical transfer functions of an immersed cylinder. The results were compared to measurements made in the free field for the same structure.

3:20

**UU2. Delay-Doppler resolution performance of large time-bandwidth product linear FM signals in a multipath ocean environment.** Jean-Pierre Hermand (SACLANT ASW Research Centre, I-19026 La Spezia, Italy) and William I. Roderick (Naval Underwater Systems Center, New London, CT 06320)

Active sonar systems that transmit large time-bandwidth (TW) product linear frequency-modulated (LFM) waveforms and receive echoes from targets of unknown speed can suffer considerable correlation losses

that cannot be predicted from conventional (narrow-band) ambiguity function theory. As is well known, the theory can be modified to include the effects of Doppler distortion on large TW-product signals by correlating the received signal against a reference that is a time-compressed version of the transmitted signal. In this paper, the effects of multipath (or target highlight structure) and Doppler on the correlation process for rectangular weighted large TW-product LFM waveforms are examined. When the received echo contains no multipath, the correlator peak output is a maximum for the reference channel that is closest in Doppler to the target. However, in a multipath environment, the correlator output peak does not generally occur at the correct Doppler reference channel. The spreading of the cross-ambiguity function as a result of Doppler mismatch and multipath is calculated in terms of the delay-Doppler coordinates and magnitude of the extrema. It is shown that, under multipath conditions, correct estimates of target parameters can be recovered through integration of the correlator output.

3:35

**UU3. Effects of correlated noise on broadband conventional and modal matched-field processing.** G. B. Smith, D. R. Del Balzo, and C. Feuillade (Naval Ocean Research and Development Activity, NSTL, MS 39529)

In a recent paper [J. Acoust. Soc. Am. Suppl. 1 **82**, S73 (1987)], the effect of realistic ambient noise on narrow-band matched-field processing in both hydrophone space and mode space was examined. A significant result of that study was that correlated noise degraded the performance of the matched-field processor in hydrophone space much more than it did in mode space. In this paper, the previous work is extended to broadband in both hydrophone space and mode space. Using computer simulations, the performance of broadband conventional and broadband modal matched-field processing is compared at various bandwidths and various mixtures of correlated and white components in the ambient noise field. Since the averaging of correlated noise (which causes the modal processor to perform better than the conventional processor in realistic noise fields) depends on the number of propagating modes, the effect of varying the center frequency of these processors is also examined.

**UU4. Matched-field processing and the effect of realistic noise correlation models.** Cedric A. Zala (Barrodale Computing Services, Suite 200, 1677 Poplar Avenue, Victoria, British Columbia V8P 4K5, Canada), John M. Ozard (Defence Research Establishment Pacific, FMO, Victoria, British Columbia V0S 1B0, Canada), and Fausto Milinazzo (Royal Roads Military College, FMO, Victoria, British Columbia V0S 1B0, Canada)

In simulating acoustic source localization by matched-field processing, it is desirable to use a realistic ambient noise model. Two general classes of noise model were considered: a planar isotropic noise source distribution and a line source distribution. The former was implemented based on the work of Kuperman and Ingenito [J. Acoust. Soc. Am. 67, 1988–1996 (1980)]. Their expressions for the correlation integrals were adapted to compute the correlation function for both simple and realistic propagation models, through the use of analytical and numerical Green's functions, respectively. The line source model involved the assumption of uncorrelated noise sources along an infinite line and was designed to model a shear zone in a shallow ice-covered ocean. Using the corresponding Green's functions for a waveguide, the line model correlation function could be reduced to a summation over all mode pairs, each term of which involved an integration over wavenumber. In this paper, the line source correlation model is presented and correlation functions given by both noise models are illustrated. The effect of using these models on the localization of a discrete source by a grid search-optimization procedure is then examined.

## 4:05

**UU5. The sensitivity of experimental, shallow-water, matched-field localization estimates to variation of environmental parameters.** C. Feuillade, W. A. Kinney, and D. R. Del Balzo (Naval Ocean Research and Development Activity, NSTL, MS 39529)

In two recent papers [J. Acoust. Soc. Am. Suppl. 1 80, S113 (1986)], the results of simulation studies to determine the effects of environmental mismatch on the robustness of matched-field processors in shallow water were studied. In this paper, the knowledge gained from those studies is applied to the interpretation of a vertical array shallow-water, matched-field experiment off Panama City, FL. How the ability to localize in range and depth is dependent upon the following factors is discussed: water depth, sound-speed discontinuity at the water/sediment interface, and sediment density and attenuation. Also discussed will be the fact that the results indicate: (a) the limitations of single-frequency, time-independent, matched-field processing and (b) the need for broadband and time-dependent techniques.

## 4:20

**UU6. Noise and array configuration considerations for matched-field processing in phone and mode space.** G. M. Fricter and D. R. Del Balzo (Naval Ocean Research and Development Activity, NSTL, MS 39529)

The conditions under which mode space matched-field processing offers performance gains relative to phone space matched-field processing are investigated. Particular emphasis is placed on determining the effects

of spatially correlated versus spatially uncorrelated noise fields and choice of array configuration on signal detection. Varying amounts of homogeneous white noise and spatially correlated noise as calculated by a normal mode noise model were combined with the field due to a submerged source to produce simulated phone space covariance matrices for a range of array configuration parameters. A phone-to-mode spacing mapping was then used to obtain the corresponding cross-amplitude correlation matrices. The results of both conventional and MLM processing were analyzed and the trade-off between the number of phones and percent of the water column spanned was investigated.

## 4:35

**UU7. A new method of ocean acoustic tomography—"Phase tomography."** E. C. Shang (Wave Propagation Laboratory, NOAA, Boulder, CO 80303)

The ray travel time has been used as the basic measurement for inversion of the sound speed structure in the conventional ocean acoustic tomography. There are cases where the ray theory is neither applicable nor convenient, especially for lower frequency and axial propagation in the ocean sound channel. In the present paper, a new method of ocean acoustic tomography has been developed; it is a full-wave approach based on the adiabatic normal mode theory. Instead of the ray travel time, the modal phase difference perturbations are used for inversion—we call it "phase tomography." The method is suitable for lower frequency, and it is also suitable for axial propagation; moreover, the range-dependent sound-speed structure can be obtained from the vertical slice inversions. The mathematical derivation will be presented and some possible ways of implementation will be discussed.

## 4:50

**UU8. Remote sensing of ocean flows with spatial filters.** Gregory B. Crawford (CIRES, University of Colorado/NOAA, Boulder, CO 80309), Richard J. Lataitis, Steven F. Clifford (Wave Propagation Laboratory, NOAA/ERL/WPL, R/E/WP, 325 Broadway, Boulder, CO 80303), and David M. Farmer (Institute of Ocean Sciences, 9860 W. Saanich Road, Sidney, British Columbia V8L 4B2, Canada)

The complicated scintillation pattern observed when an acoustic wave propagates through refractive index irregularities is due to the "projection" of all the refractive eddies at each path position onto the receiving plane. A current transverse to the acoustic path will advect the eddies and produce a drift in the scintillation pattern. A single transmitter and two receivers can be used to monitor the drift and infer a path-averaged flow speed. Much more localized measurements of the transverse current can be made with multiple sources and receivers that spatially filter the scintillation pattern. Such spatial filter systems can be used to profile the transverse current. Recent results obtained with small transmitter and receiver arrays mounted across a well-mixed, turbulent tidal channel are presented.

## Meeting of Accredited Standards Committee S1 on Acoustics

D. Johnson, Chairman S1

*Larson-Davis Laboratories, 280 South Main, Pleasant Grove, Utah 84062*

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

The international activities in ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and IEC/TC 87 Ultrasonics will be discussed. The Chairs of the respective U. S. Technical Advisory Groups for ISO/TC 43 (H. E. von Gierke), IEC/TC 29 (V. Nedzelnitsky), and IEC/TC 87 (P. D. Edmonds) will report on current activities of these Technical Committees.

**Session VV. Engineering Acoustics V: General Engineering Acoustics**

Steven C. Thompson, Chairman

*Acoustics and Sensor Systems, Ocean Systems Division, Gould Inc., 18901 Euclid Avenue, Cleveland, Ohio 44117***Contributed Papers****8:00**

**VV1. The attenuation of dissipative flow duct mufflers with internal mean fluid flow in the absorbent.** A. Cummings (Department of Engineering Design and Manufacture, University of Hull, Hull HU6 7RX, United Kingdom)

In dissipative flow duct mufflers incorporating layers of porous bulk sound absorbent, mean flow within the pores of the material is inevitable because the porous medium is exposed to pressure gradients in the duct flow. The acoustical effects of internal mean flow can be considerable, particularly in the case of internal combustion engine exhaust mufflers of the dissipative type. Theoretical treatments are presented here, both to describe the effect of internal flow on the bulk acoustic properties of the porous medium and to find the effect of the absorbent *in situ*, in the form of the sound transmission loss of the silencer. The measured transmission loss of an experimental silencer is compared to predicted data, and good agreement between the two is obtained. The effects of mean fluid flow in the central passage and internal flow in the absorbent are separately demonstrated.

**8:15**

**VV2. Perturbations of the acoustic field inside a resonant cavity.** Philippe Herzog and Michel Bruneau (Laboratoire d'Acoustique, Université du Maine, B.P. 535, 72017 Le Mans Cedex, France)

A theoretical and experimental study has been made of the acoustic field inside a closed resonant cavity with two axes of resonance under the influence of several perturbations, like an imperfect geometry, the presence of the transducers, and the effects of the motion of the cavity (linear and angular speeds and accelerations). The analytical study is based on a modal decomposition of the pressure, for Cartesian and cylindrical shapes. The influence of geometrical perturbations and the presence of the transducers are taken into account by a perturbation method applied to a matrix formulation of the Helmholtz-Huygens equation. The influence of inertial terms is expressed in a similar form, but one which may be used for greater mode coupling. The experimental work has been done in the Cartesian case for a two-dimensional cavity. The use of a swept-sinusoidal excitation with synchronous demodulation and a custom-made acoustic antenna permitted us to obtain very accurate measurements. Several examples are shown, and the agreement with theory is quite good, leading to discrepancies typically less than 0.1 dB and the detection of modes in a dynamic range of about 50 dB.

**8:30**

**VV3. Scattering from thin elastic screens.** Christian Audoly (Groupe d'Etudes et de Recherche de Detection Sous-Marine, Le Brusac, 83140 Six-Fours, France)

The characteristics of acoustical materials are often determined by standard panel tests. To improve low-frequency measurements, the scattering effects due to the finite size of the panels are investigated. A theoretical model, based on the Helmholtz integral equation and its derivative form, is used to calculate the scattering of acoustic waves from rigid, soft,

and elastic screens in water. The cases of the infinite strip (2-D space) and of the rectangular plane screen (3-D space) are considered. In both cases, it is assumed that the thickness of the screen is small compared to the wavelength. The calculations are made up to  $ka = 10$  ( $k$ , wavenumber;  $a$ , characteristic dimension). The results are compared to experiments on soft material and steel screens, and to the infinite plane model. It is shown that in this frequency range, the scattering effects due to the finite size of the screens are important, mainly for the steel panel.

**8:45**

**VV4. Ultrasonic tomography in the field of flow measurement.** Jörg Wolf (Institut für Mess- und Regelungstechnik, Universität (TH) Karlsruhe, Postfach 6980, D-7500 Karlsruhe 1, Federal Republic of Germany)

Ultrasonic tomography is known as a nonintrusive imaging method for static sound attenuating objects in fields such as medical diagnostics or nondestructive testing. A new application of ultrasonic tomography in chemical engineering is presented in this paper, namely, the nonintrusive imaging of the static fluctuating void fraction in a cross section of a gas liquid flow. Here, the attenuation of parallel ultrasonic rays is measured. The set of values obtained from this measurement result in a projection of the flow. Now the spatial distribution of void fraction can be reconstructed from a set of these projections, taken from various directions, by the algorithms of computerized tomography [Proc. IEEE, 21(3) (1983)]. The method was applied to a bubble column as used in chemical engineering. To get a fast and nondisturbed measurement of the projection data, a circular array of 108 piezoceramic transducers was directly integrated in the pipe wall of the column. The transducers are electronically switched and need not be mechanically operated. The control of the array and the reconstruction algorithms are implemented on a microcomputer coupled with an array processor.

**9:00**

**VV5. Scattering from slots.** Marian Smith (Lockheed Palo Alto Research Laboratory, O/91-60, B/256, 3251 Hanover Street, Palo Alto, CA 94304)

The effect of slots in boundaries on scattered radiation is a classical problem, yet one which is still of considerable interest. This paper presents an extension of prior work by the author [J. Sound Vib. (to be published July 1988)], which considered the effect of covering only the long wavelength approximation, where the slot width was much smaller than the wavelength of the incident radiation. This paper presents a more general method for the calculation of scattering from slots, which allows for solutions for larger slots, limited only by resonance considerations and computer size. Examples of solutions for scattering from slots in rigid walled plates, ducts, and cavities are presented. The method reduces the scattering problem, by means of the Green's function techniques, to an integral or integrodifferential equation. The resulting equation can be solved by numerical methods.

9:15

**VV6. Laser Doppler anemometry detection of hydroacoustic particle velocity.** P. S. Dubbleday and H. C. Schau (Underwater Sound Reference Detachment, Naval Research Laboratory, P.O. Box 568337, Orlando, FL 32856-8337)

The method of Laser Doppler Anemometry (LDA) is applied to the measurement of particle velocity in a hydroacoustic field. The field is created in a USRD type G19 or G40 calibrator. The LDA results are compared with the values for the velocity obtained from the pressure gradient measured by a USRD standard hydrophone. The anemometer is operated in forward scattering mode, and also in backscattering mode through the use of a fiber-optic transducer. Seeding particles of varying material and size are dispersed in the acoustic medium (water). The influence of these parameters including the particle concentration on the signal quality and detection threshold is discussed. [Work supported by ONR.]

9:30

**VV7. Transmission loss of a piping network.** D. C. Stredulinsky, A. Craggs, and M. G. Faulkner (Department of Mechanical Engineering, University of Alberta, Edmonton, Alberta T6G 2G8, Canada)

This paper describes a numerical procedure developed to analyze piping and duct systems. The procedure combines transfer matrix and finite element methods and was used to predict the transmission loss of a piping network constructed of rigid-walled, square cross-section tubing. The network consisted of one input, one output, and several segments made up of straight pipes and 90° elbows. These were interconnected at several "T" junctions. The junctions and elbows were treated using finite element models that took into account two-dimensional wave propagation. These were connected to straight pipes in which one-dimensional wave propagation was assumed. This led to a much reduced connective matrix for each finite element model. The solution of the entire network was then obtained

10:15

**VV9. Influence of ground reflection on measurements involving bands of noise.** John M. Noble, Henry E. Bass, and Richard Raspet (National Center for Physical Acoustics, P.O. Box 847, University, MS 38677)

It has been shown [L. C. Sutherland and H. E. Bass, *J. Acoust. Soc. Am.* **66**, 885-894 (1979)] that frequency-dependent atmospheric absorption can lead to a propagation loss for a band of noise that is much different from that for a pure tone at band center. In the presence of a ground surface, interference can also cause the sound amplitude to vary rapidly with frequency. When this occurs, the level measured for a pure tone can differ dramatically from that measured for a band of noise. Accurate treatment of this difference requires integration over the bandpass of the fractional octave band filter used in the measurement. Example calculations have been performed for a typical filter. These examples form a basis for general guidelines to be used when comparing theory to measurements.

10:30

**VV10. Audio processing for improved detection and recognition of threats in radar warning systems.** Charles H. Wiseman (Lockheed Missiles & Space Company, Inc., Astronautics Division, Advanced Marine Systems, P.O. Box 3504, Sunnyvale, CA 94088-3504)

by solving a relatively small system of linear equations. Experimental measurements of the network transmission loss were made using a two-microphone transfer function technique with random "white" noise and a dual channel FFT analyzer. The resultant sound transmission loss was in good agreement with the predicted transmission loss up to the cut-on frequency of cross modes within the straight pipe sections.

9:45

**VV8. The use of numerical optimization techniques in the design of acoustic horn loudspeakers.** James McLean (TRW Antenna Systems Laboratory, Redondo Beach, CA 90277) and E. L. Hixson (The University of Texas at Austin, Austin, TX 78712)

The use of numerical optimization techniques in the design of acoustic horn loudspeaker systems is considered. Conventional synthesis techniques (i.e., stepped impedance transformer or nonuniform line) are inadequate for horn loudspeaker design due to the frequency-dependent nature of the radiative load at the mouth and the complex behavior of the driver. In this paper, the horn is modeled as an  $n$ -section commensurate stepped impedance transformer terminated in a one-port network that represents the complex frequency-dependent radiation impedance. The driver is modeled as a linear two-port transducer using measured data as described in an earlier presentation [J. S. McLean and E. L. Hixson, *J. Acoust. Soc. Am. Suppl.* **1 79**, S90 (1986)]. The entire horn/driver system is optimized to exhibit equiripple frequency response using the characteristic impedances of the individual transformer sections as the optimization variables. The system is optimized as a singly terminated filter in order that it may be driven by a low-impedance power amplifier. The optimized horn/driver system exhibits a bandpass response. This differs from previous horn designs which were high pass in nature. In the optimized design, the bandwidth of the system may be increased by allowing a higher ripple level and vice versa.

10:00-10:15

Break

Radar warning receivers (RWR) are systems that alert pilots of military aircraft to imminent danger of attack by missiles and gunfire. These RWRs function to: (a) intercept radar transmissions; (b) detect the radar's pulsed modulations; (c) identify the patterns of pulses as emanating from friend or foe, and, finally, (d) present information as to threat type and direction via alpha numerics and symbols on a video display and, aurally, via a headset. The aural presentation termed "raw audio" is entirely an analog process. It is used to complement the video display as an aid in the detection and recognition of threats. In order to decrease clutter and distraction to the pilot, modern RWRs incorporated computer analysis to assist in discerning threats from the myriad of PRF patterns. With computer logic, of course, the demodulated pulse data must first be converted to digital form. This process, however, has eliminated the analog raw audio in modern RWRs in favor of "synthetic audio," reconstituted from digital data. This paper discusses the differences between raw audio and synthetic audio and their advantages and disadvantages. The paper concludes that computer-driven RWRs supplemented with raw audio capability show considerable promise for improved detection and recognition capability and should be expeditiously investigated.

10:45

**VV11. Sum and difference frequency generation in sound radiation from a vibrating planar boundary.** Mosaad A. Foda (Faculty of Engineering, El-Mansoura University, Egypt)

An earlier study [J. H. Ginsberg, J. Acoust. Soc. Am. 69, 60-65 (1981); 69, 929-936 (1981)] of sound radiation from a vibrating planar boundary considered the case of monochromatic excitation. That work featured perturbation analysis combining asymptotic integration and the renormalization version of the method of strained coordinates. The present analysis initiates an extension of those techniques to the case of a two-frequency excitation. These frequencies are considered arbitrary; hence, the results for a parametric array (closely spaced frequencies) can be

obtained as a special case of the present analysis. Using the Fourier transform in its conventional complex function form to describe the linearized signal leads to formulation of the second-order potential in terms of complex functions. A coordinate straining transformation describing the full spectrum (propagating and evanescent) is deduced. Accordingly, uniformly accurate expressions for the acoustic pressure and the velocity components are obtained.

FRIDAY MORNING, 20 MAY 1988

EAST BALLROOM A, 8:10 TO 11:30 A.M.

### Session WW. Education in Acoustics IV and Physical Acoustics VIII: Computer-Based Theoretical Studies in Acoustics

Charles Thompson, Chairman

*Department of Electrical Engineering, University of Lowell, 1 University Avenue, Lowell, Massachusetts 01854*

Chairman's Introduction—8:10

#### *Invited Papers*

8:15

**WW1. MACSYMA automated symbolic mathematics software.** Richard Petti (Symbolics Inc., Cambridge, MA 02139)

Just as computers revolutionized numerical analysis, so symbolic mathematics software is revolutionizing symbolic mathematics computations, yielding major increases in speed accuracy and modeling power in many applications. In addition to automating large and tedious computations, MACSYMA provides automated mathematical expertise in many areas, such as solution of simultaneous equations, integration, Laplace transforms, ordinary differential equations, and simplifications of expressions. The presentations will focus on selected capabilities of MACSYMA that are relevant for acoustics, especially: (1) symbolic vector and tensor calculus, (2) symbolic solution of ordinary differential equations, (3) symbolic approximation methods, (4) use of symbolic math software to generate numerical analysis code, (5) symbolic matrix computations. A partial bibliography of research papers in fluid mechanics reporting on work where MACSYMA was used will also be provided. Also discussed will be the role of symbolic mathematics software in technical education.

8:45

**WW2. Scratchpad II: A computer algebra language and system.** Richard Jenks (T. J. Watson Research Center, Yorktown Heights, NY 10598)

The Scratchpad II system represents a new generation of systems for doing symbolic mathematics, based on modern algebra and abstract data types. A large number of facilities are provided, for example: symbolic integration, "infinite" power series, differential operators, Cartesian tensors, and solution of nonlinear systems. Scratchpad II has been designed from the outset to be extendible. The system introduces a new data abstraction notion, the "category," to express intricate interrelationships between data types. The result design permits the compilation of algorithms described in their most natural mathematical setting. The use of categories guarantees user defined types and packages are compatible with each other and with built in facilities. This system provides a single high-level language with an interpreter and compiler. The language can be used by the naive user for convenient interactive mathematics calculations and by the advanced user for the efficient implementation of algorithms. Scratchpad II is built on Lisp/VM and runs on IBM/370 class mainframes. An implementation of the system on the RT/PC is expected soon.

9:15

**WW3. A survey of FORTRAN code generation using MACSYMA.** Stanly Steinberg (Department of Mathematics and Statistics, University of New Mexico, Albuquerque, NM 87106) and Patrick J. Roache (Ecodynamics Research Associates Inc., Albuquerque, NM 87106)

This paper presents the underlying symbolic manipulation techniques used by the authors to produce FORTRAN code with the artificial intelligence code MACSYMA. The FORTRAN codes produced use finite difference methods to solve the governing equations for numerical grid generation, including elliptic and variational methods, and to solve the hosted equations, including fluid dynamics, heat transfer, and electro-statics. Consideration will be given to writing the hosted equations in strong conservation form in general nonorthogonal coordinates; high order and conditional differencing schemes; colocated variables versus staggered grids; and code validation procedures.

9:45

**WW4. Use of symbolic manipulation in physical acoustics.** Martin Manley and Vineet Mehta (Electrical Engineering Department, University of Lowell, 1 University Avenue, Lowell, MA 01854)

The development of a mathematical theory in acoustics (or in any other field) depends on the insight of the theoretician. However, the formulation of an insight into an equation derived from first principles involves much time spent in activities that fall within the category of a symbolic manipulation. To demonstrate the role symbolic manipulation software can play in the solution process, classical problems in acoustics will be selected. It is known that the solution to the linearized governing equations of an acoustic field can be considered as a superposition of vorticity, thermal, and acoustic modes. However, these modes are coupled. The separation of these modes will be demonstrated. Symbolic manipulation will be used to explore how the higher-order terms in the perturbation parameters,  $\epsilon_\mu$  and  $\epsilon_\nu$ , affect the modal coupling. The issue of asymptotic degeneracy will also be addressed (the effects of varying the rates at which these parameters approach zero).

10:15

**WW5. An evaluation of an intelligent computer service assistant.** Paul W. Armstrong and Dan LaRue (Apollo Computer, Inc., Chelmsford, MA 01824)

This presentation will discuss the development issues required for the implementation of an intelligent assistant for Apollo's customer service organization. An overview of the design and requirement goals as well as the integration with current data systems and input will be addressed. Maintenance and future enhancements will be reviewed as well as a discussion of the user interface and user profiles.

### *Contributed Papers*

10:45

**WW6. Application of symbolic computation to singular perturbation problems in physical acoustics.** Charles Thompson (Department of Electrical Engineering, University of Lowell, 1 University Avenue, Lowell, MA 01854)

Problems in fluid mechanics, and in turn in physical acoustics, often incorporate a large number of length scales and parameters. These scales are the result of local fluid interactions with either boundary surfaces or across discontinuities in the physical properties of the fluid. The art of developing a correct description of any physical situation is in discerning the relative importance of each of the parameters of the solution. One might consider solutions to a problem as a restricted range of values of these parameters. In such a case, these solutions would be locally valid in the parameter space and are an asymptotic representation of the global solution. The integration of locally valid solutions into a global description is the essence of singular perturbation methods. In this paper, the method by which computer algebra can be used to solve problems involving a singular perturbation analysis in physical acoustics will be discussed.

11:00

**WW7. Creation of experiment-oriented materials to enhance a nonlab acoustics elective.** S. A. Elder and M. S. Korman (Physics Department, U.S. Naval Academy, Annapolis, MD 21402)

Two acoustics courses are offered by the Naval Academy Physics Department. One is a physics major elective course in physical acoustics with a weekly 2-h lab; the other, a sonar course, is basically a service course for other departments. On account of the large attendance for the latter, and the limited space available, it has not been possible to make lab work a routine part of the course. Since it is believed that some lab experience is essential to good physics teaching, materials are being created to be used by students outside of class time that will address this need. Among ideas being considered are computer-graphic homework assignments, special project-type lab setups, video-cassette instructional aids, and sonar games. Current materials and experience will be described.

11:15

**WW8. Farfield criteria for vibrating structures: Numerical simulation on plates and cylinders.** Christophe Journeau (Consulting Engineer for Centre d'Etudes et de Recherches pour la Discrétion des Navires, DCAN Toulon, 83800 Toulon Naval, France), Thierry Rohan, and François Pellegrini (Centre d'Etudes et de Recherches pour la Discrétion Acoustique des Navires, DCAN Toulon, 83800 Toulon Naval, France)

In order to determine practical criteria for the nearfield/farfield transition in the case of underwater vibrating structures, the acoustic field radiated by plates and cylinders is simulated. Various mode shapes of



these structures are computed for different frequencies including their resonance frequencies. These calculations are made for radiators of different sizes and are compared with the results of a mechanical simulation for a point-excited ribbed cylinder. Then the distances at which the pressure field is at  $\pm 1$  dB of the farfield value, the wave specific impedance is

$\rho c \pm 12\%$  (1 dB), and pressure and velocity have a phase difference of less than  $\pm 15^\circ$  are tabulated as a function of frequency. The envelope of these points is compared with the standard criterion ( $R \gg \lambda$  and  $R \gg L^2/\lambda$ ). Finally, various criteria are proposed depending on which property of the farfield is required.

FRIDAY MORNING, 20 MAY 1988

WEST BALLROOM A, 8:15 A.M. TO 12:00 NOON

## Session XX. Physical Acoustics IX: Cavitation, Bubbles, and Medical Acoustics

Ronald A. Roy, Chairman

Department of Mechanical Engineering, Yale University, Box 2159, Yale Station, New Haven, Connecticut 06520

### Contributed Papers

8:15

**XX1. The Blake threshold of a bubble having a radius-dependent surface tension.** Anthony A. Atchley (Physics Department Code 61 Ay, Naval Postgraduate School, Monterey, CA 93943)

The Blake threshold is incorporated in most theoretical models of transient acoustic cavitation. In standard derivations, this threshold is calculated for a bubble having constant surface tension. However, such a bubble is unstable against dissolution and not a realistic model for a cavitation nucleus. In this work, the Blake threshold is calculated for a more realistic nucleus—a bubble having a radius-dependent surface tension. A relationship is derived for this dependence, based upon the properties of a surfactant-stabilized cavitation nucleus. Two cases, appropriate for charged and polar surfactants, are examined. The Blake threshold for the former case is identical to that of the standard case by approximately 2%–20% for radii ranging from  $3.0 \mu\text{m}$  down to  $0.01 \mu\text{m}$ . Therefore, it is concluded that both unstabilized and surfactant-stabilized bubbles become mechanically unstable under essentially the same acoustic conditions. Thus the use of the simpler, unrealistic nucleus in theoretical models results in no serious error. [Work supported by NPS Foundation.]

8:30

**XX2. Measurement of acoustical cavitation in liquid  $^4\text{He}$ .** Joel A. Nissen, E. Bodegom, L. C. Brodie, and J. S. Semura (Department of Physics, Portland State University, Portland, OR 97207)

A piezoelectric hemispherical transducer was used to focus high-intensity ultrasound into a small volume of liquid helium. The transducer was gated at its resonant frequency of 566 kHz, with gate widths less than 1 ms, in order to minimize the effects of transducer heating and acoustic streaming. The onset of cavitation was detected with small-angle scattering of laser light from the cavitation zone by microscopic bubbles. In the superfluid, the pressure amplitude at the focus was calculated based upon the acoustic power radiated into the liquid, the geometry of the transducer, and the nonlinear absorption of sound. Above the lambda point, this method is complicated by the scattering of sound by bubbles present in the liquid and on the surface of the transducer, and so a calibration curve for the diffracted light intensity at different pressure amplitudes and temperatures was obtained in the superfluid. The diffracted light intensity was then used to determine the acoustic pressure and extend the cavitation measurements to the normal fluid. [Work supported by NSF.]

8:45

**XX3. An active acoustic backscattering technique for detecting transient cavitation.** Ronald A. Roy, Sameer Madanshetty, and Robert E. Apfel (Department of Mechanical Engineering, Yale University, Box 2159 Yale Station, New Haven, CT 06520)

Relatively dirty water is insonified with short pulses (order  $10 \mu\text{s}$ ) of high-frequency ( $\approx 1$  MHz), focused ultrasound. The subsequent transient cavitation bubble activity is detected by a focused 30-MHz pulse-echo detector confocally positioned relative to the insonifying transducer. The cavitation detector is sensitive enough to observe submicron size bubbles. Transient cavitation thresholds measured using this active detection system are as much as 40% lower than those measured using other techniques (such as passive acoustic detection). Thresholds are measured as a function of various acoustic parameters (e.g., pulse length, pulse repetition frequency, etc.) and the results are compared with previous investigations as well as theoretical predictions. [Work supported by NIH Grant No. 1 R01-CA-39374.]

9:00

**XX4. Brewster angle light scattering from bubbles in water: Observations and potential applications to the acoustics of natural microbubbles.** S. M. Bäumer and P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164-2814)

Consider the reflection of light from a clean gas bubble in water such that the light is polarized with its electric field parallel to the scattering plane. The Fresnel reflection coefficient should vanish when the local angle of incidence  $i$  is at Brewster's angle  $i_B = \arctan(n^{-1}) \approx 36.9$  deg, where  $n = 1.33$  is the refractive index of water. When the scattering angle  $\theta$  is close to that of the Brewster scattering angle,  $\theta_B = (180 - 2i_B) \approx 106.2$  deg, this Brewster effect should be strongly manifested in the scattering pattern [D. L. Kingsbury and P. L. Marston, *Appl. Opt.* **20**, 2348–2350 (1981)]. In the present research the first laboratory observations described are of Brewster effects in the scattering patterns of bubbles rising through water having radii  $\approx 70 \mu\text{m}$ . There is a noticeable reduction in the visibility of fringes in the scattering pattern for scattering angles  $\approx \theta_B$ . (The fringes arise from the interference of a far-side ray with the reflected ray.) Microbubbles of natural origin can be important in ocean and bioacoustics and Brewster effects may be useful in the optical characterization of such bubbles including the discrimination

of optical scattering patterns of bubbles from those of particles or cells. Brewster angle scattering may also be useful for characterization of the surfactant coating that is thought to occur on some microbubbles in nature. [Work supported by ONR.]

9:15

**XX5. Enhanced nonlinearity in a bubbly liquid considered as a mixture.** Erich Carr Everbach and Robert E. Apfel (Yale University, 2159 Yale Station, New Haven, CT 06520)

During the past decade there has been an increasing interest in the nonlinear behavior of gas bubbles in liquids. Recent papers by Soviet investigators [e.g., Kustov *et al.*, *Sov. Phys. Acoust.* 32(6), 500-503 (1986)] have concentrated on phenomena observed when sound interacts with a bubbly layer. These phenomena can include acoustic solitons, parametric radiation, self-focusing, phase conjugation, and other effects that occur because the system is highly nonlinear. An investigation of the acoustic nonlinear parameter of mixtures was conducted to help infer the volume fractions of the mixture components. When a bubbly medium of known air volume fraction is considered as a two-component mixture, its effective nonlinear parameter may be calculated using mixture methodologies [R. Apfel, *J. Acoust. Soc. Am.* 74, 1866-1869 (1983)]. The mixture analysis in this work, which is independent of any assumptions regarding bubble oscillation, yields nonlinear parameter values comparable to those of the Soviet investigators. For certain volume fractions, values of the nonlinear parameter of a bubbly liquid may be hundreds of times that of either the gas or the liquid host. Both experimental and theoretical work will be presented and the implications of this enhanced nonlinearity will be discussed. [Work supported by NIH through Grant 5-R01-GM30419.]

9:30

**XX6. Measurement uncertainty assessment of the scanning laser acoustic microscope and application to canine skin and wound.** William D. O'Brien, Jr., Dianne L. Steiger (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801), John E. Olerud, Mary Ann Riederer-Henderson, and George F. Odland (Department of Medicine, University of Washington, Seattle, WA 98195)

The assessment of measurement uncertainty of the scanning laser acoustic microscope (SLAM) has not been thoroughly evaluated experimentally. It is essential that the uncertainty of any measurement be fully understood. Both accuracy and precision were experimentally evaluated for the SLAM by measurements on homogeneous liquids of known ultrasonic properties. Using aqueous solutions of bovine serum albumin, the attenuation coefficient accuracy and precision are  $\pm 12\%$  and  $\pm 15\%$ , respectively. And from Dow Corning 710, a silicon oil, the speed accuracy and precision are  $\pm 2.9\%$  and  $\pm 0.4\%$ , respectively. Further, an application of the assessment of precision was conducted using duplicate samples of dog skin and wound tissue. From these evaluations of a heterogeneous tissue, the estimated precision in the measurement of the attenuation coefficient and speed was  $\pm 16\%$  and  $\pm 1.7\%$ , respectively. [Work supported by NIH Grant AM 21557 and the Rehabilitation Research and Development Service of the Veterans' Administration.]

9:45

**XX7. Propagation of ultrasound in skin.** Mary Ann Riederer-Henderson (1057 Summit Avenue East, Seattle, WA 98102), John E. Olerud, George F. Odland (Department of Dermatology, University of Washington, Seattle, WA 98195), Fred K. Forster (Department of Mechanical Engineering, University of Washington, Seattle, WA 98195), Dianne L. Steiger, and William D. O'Brien, Jr. (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801)

The scanning laser acoustic microscope (SLAM) at 100 MHz and backscattering acoustic technique (BAT) at 10-40 MHz were used to examine canine skin. Specimens from four animals and from four locations on the animal were analyzed biochemically and morphologically as well as acoustically. The mean collagen and water concentrations were  $20\% \pm 2\%$  and  $60\% \pm 3\%$ , respectively. An analysis of variance of the SLAM data and the biochemical data showed significant animal to animal differences and some differences due to location on the animal. At 100 MHz the mean ultrasonic speed obtained with the SLAM was  $1632 \pm 34$  m/s and the mean attenuation coefficient was  $57 \pm 10$  dB/mm. Using BAT the mean integrated attenuation coefficient was  $11 \pm 3$  dB/mm at 25 MHz (midfrequency range). While the speed values fall within the range of values previously reported for skin, the values for the attenuation coefficient using either SLAM or BAT are considerably higher than would be predicted from literature values at 1-10 MHz. Thus the attenuation coefficient is a stronger function of frequency than the data at lower frequencies would suggest. [Work supported by NIH and RR&D of the Veterans Administration.]

10:00

**XX8. Off-resonance contributions to acoustic backscattering from bubble swarms.** Kerry W. Commander and Elan Moritz (Naval Coastal Systems Center, Physical Acoustics Branch, Code 4120, Panama City, FL 32407)

Recent discrepancies between measurements of bubble populations near the surface in the ocean by optical and acoustical methods have been discussed in detail by MacIntyre [F. MacIntyre, "On reconciling optical and acoustical bubble spectra in the mixed layer," in *Oceanic Whitecaps*, edited by E. C. Monohan and G. MacNiocaill (Reidel, New York, 1986), pp. 75-94.] Another possible explanation for the very large number of small bubbles found by acoustic methods is the potential for overestimates of bubble numbers through the use of resonance scattering theory in the bubble population calculations. In this work it is shown that for some bubble distribution of interest, the acoustic backscattering for a particular frequency is actually dominated by scattering contributions from off-resonance bubbles. This effect is more prevalent for high frequencies, causing overestimation of the number of small bubbles, which is precisely where the two methods disagree. The more complete acoustic calculations may not completely reconcile the bubble number estimates obtained from the optical and acoustical bubble spectra; however, they bring them closer together. [Work supported by ONT.]

10:15

**XX9. Ultrasonic propagation properties of articular cartilage at 100 MHz.** D. H. Agemura, W. D. O'Brien, Jr. (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801), J. E. Olerud, L. E. Chun, and D. R. Eyre (University of Washington, Seattle, WA 98108)

A pilot study on articular cartilage assessed the contribution of individual matrix components to ultrasonic propagation. The influence of collagen fibril orientation and collagen crosslinking was also assessed. Sections of adult bovine articular cartilage taken both parallel and perpendicular (cross sections) to the articular surface were examined using the scanning laser acoustic microscope (SLAM) operating at an ultrasonic frequency of 100 MHz. A set of samples was evaluated that had been sequentially treated by enzymes to (1) remove 85% of the chondroitin sulfate; (2) remove remaining GAGs, glycoproteins, and other noncollagen proteins, leaving only the collagen fibril network; and (3) disrupt the collagen intermolecular crosslinks. Two striking observations were made: a profound effect of the "preferred" collagen fibril orientation on acoustic wave speed and a marked increase in attenuation coefficient when intermolecular crosslinks were broken in the collagen. [This work was supported by NIH Grants AM21557, CA36029, and AR36794 and Rehabilitation Research and Development Services of the Veterans Administration.]

10:30

**XX10. Ultrasonic propagation properties of canine aorta at 100 MHz.** Felice Chu (Bioacoustics Research Lab., Department of Electrical and Computer Engineering, University of Illinois, Urbana IL 61801), Mary Ann Riederer-Henderson, J. E. Olerud (University of Washington, Seattle, WA 98108), and William D. O'Brien, Jr. (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801)

A pilot study examining canine aorta investigated the ultrasonic propagation properties as a function of location and orientation along the aorta. Longitudinal, circumferential, intimal, and adventitial sections of canine aorta were taken at seven locations along the aorta from the aortic arch to the iliac bifurcation. Aortic media consists of circumferentially oriented elastin fibers interspersed with a randomly oriented network of fibrils. Over the distance from the heart to the abdomen, the elastin concentration decreases as the collagen concentration increases. The speed is shown to be statistically significant with respect to location and correlates strongly with orientation. The 28 speed values ranged from 1610–1859 m/s with the average being 1662 m/s. On the other hand, the attenuation coefficient is not shown to be statistically significant with respect to orientation but does correlate strongly with respect to location. The 27 attenuation coefficient values ranged from 48.5–131.8 dB/mm with an average of 91.2 dB/mm. [This work was supported by NIH Grants AM21557, CA36029, and AR36794 and Rehabilitation Research and Development Services of the Veterans Administration.]

10:45

**XX11. Ultrasonic absorption properties in excessively fatty rat liver.** Nadine B. Smith, W. D. O'Brien, Jr., and J. W. Erdman, Jr. (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801)

The ultrasonic absorption coefficient was examined in rat liver due to the effects of 1% dietary orotic acid measured using the transient thermoelectric technique @ 1MHz. Dietary orotic acid causes fat concentration to accumulate excessively in rats in a short period of time. Tissue constituents that are particularly important in absorption are water, protein, collagen, and lipid levels. The concentration of these tissues constitutes in rat liver over a 0-to 10-day period on the orotic acid diet compared to the absorption coefficient suggest that ultrasonic properties can be modeled as a function of constitute concentrations. Using the statistical analysis system, comparisons were made between the tissue constituents and ultrasonic absorption coefficient where comparisons are significant at the  $p < 0.05$  level. Lipid levels of the fatty liver were shown to be statistically significant to the absorption coefficient. Strong correlations of both collagen and protein with absorption, which have  $p$  values of 0.098 and 0.071, respectively, were observed. [Work supported by NIH Grant CA36029.]

11:00

**XX12. Accurate measurement of blood flow using ultrasound and time domain correlation.** Ilmar A. Hein and William D. O'Brien, Jr. (Department of Electrical and Computer Engineering, University of Illinois, Urbana, IL 61801)

Using time domain correlation, the volume fluid flow in a vessel can be accurately and precisely estimated without any previous knowledge of the vessel size, flow velocity profile, or transducer measurement angle. This technique observes the change in arrival time (instead of change in frequency) of ultrasound reflected from moving scatterers. If a vessel with moving fluid is insonated by two ultrasonic pulses separated by time  $T$ , the scatterers within the ultrasonic beam will move some distance  $D$  between the first and second pulse. There will be a difference  $\tau$  in the arrival times of the two pulses since the scatterers have moved. The scatterer velocity can then be calculated from the time difference  $\tau$ . If the velocity is sampled at different positions within the vessel, the volume rate of flow can be calculated, along with the vessel dimensions and transducer measurement

angle. The time domain correlation technique has been verified in a blood flow phantom system using both a blood mimicking substance and porcine blood. The transducer measurement angle has been determined within 5%, and pulsatile flow has been measured up to 150 beats/min with accuracies better than 18%. Currently, this method is being experimentally investigated under *in vivo* conditions, with the eventual application being the diagnosis of venous thrombosis in humans. [Work supported by NSF/NHLBI Grant 1 RO1 HL 37904.]

11:15

**XX13. Ultrasonic attenuation coefficient assessment in rat liver from carbon tetrachloride.** David J. Boatright, Jr., W. D. O'Brien, Jr. (Department of Electrical and Computer Engineering), and J. W. Erdman, Jr. (Department of Food Science, University of Illinois, Urbana, IL 61801)

The ultrasonic attenuation coefficient is examined in rat liver due to the effects of inhalation of carbon tetrachloride using the insertion-loss technique @ 1.385, 4.210, 7.015, and 9.820 MHz; it is theorized that rat livers will develop an increase of collagen. Tissue constituents of glycogen, protein, and water concentration are assessed. The concentration of these tissue constituents over an 11-week period on the carbon tetrachloride compared to the attenuation coefficients at the four frequencies suggest ultrasonic properties can be modeled as a function of constitute concentrations. At 1.385 MHz, the attenuation coefficient is statistically significant as a function of glycogen, water, and protein. At 4.210 MHz, only protein is statistically significant; at 7.015 MHz, both glycogen and protein are statistically significant; and at 9.82 MHz, none of the tissue properties is shown to be significant with respect to the attenuation coefficient. [Supported by NIH Grant CA 36029.]

11:30

**XX14. Wideband acoustic microscopy of tissue.** Christopher M. W. Daft (Bioacoustics Research Laboratory, Department of Electrical and Computer Engineering, University of Illinois, 1406 W. Green, Urbana, IL 61801) and G. A. D. Briggs (Department of Metallurgy, Oxford University, Parks Road, Oxford OX1 3PH, United Kingdom)

A scanning acoustic microscope (SAM) has been used to measure the elastic properties of tissue with a resolution of around  $8 \mu\text{m}$  (C. M. W. Daft *et al.*, Proc. 1986 IEEE Ultrasonics Symposium, 945–948). This is achieved by broadband excitation of the acoustic lens, and the recording of an undemodulated returning signal. This has the effect of separating the various contributions normally present in a SAM image of tissue. A method of analyzing this information to yield the sound velocity, acoustic impedance, section thickness, and acoustic attenuation is described. Results are presented from a sample of skin tissue, and sources of experimental error are discussed. The elastic properties of tissue appear to vary more substantially on this scale than in macroscopic investigations. High values of acoustic velocity and attenuation are observed in areas with high concentrations of structural protein and low water content. [Work supported by SERC and GEC Hirst Research Centre.]

11:45

**XX15. Acoustical quantification of thermal injury.** Anne W. Holmes (Center for Bioengineering, University of Washington, Seattle, WA 98195), Fred K. Forster (Department of Mechanical Engineering, University of Washington, Seattle, WA 98195), John E. Olerud (Department of Dermatology, University of Washington, Seattle, WA 98195), and Sam R. Sharar (Department of Surgery, University of Washington, Seattle, WA 98195)

In this study, acoustic imaging was augmented by tissue characterization to more accurately quantify severity of thermal injury. The acoustic

attenuation coefficient in normal and burned skin was investigated over the range of 10–30 MHz. Fourteen normal and 21 full-thickness scald burns (1 and 2 min) from goats were analyzed. The results of this study show the importance of choosing sites for attenuation estimates from the image since both normal and burned skin cannot be assumed homogeneous. From regions chosen in the reticular dermis, it was found that the attenuation coefficient was significantly less for the burns. After averaging the results from each group, the integrated attenuation coefficient over the frequency range studied was 8.3 Np/cm for normal skin, and de-

creased approximately 30% to 5.7 Np/cm for burns. The differences between normal and burned skin were also characterized by the frequency dependence of the attenuation with a power law of the form,  $af^b$ . Characterized in this manner, the fitting parameters exhibited large changes when comparing values for normal and burned skin. The parameter  $a$  increased by more than a factor of 5 and the exponent  $b$  decreased by approximately a factor of 2. These quantitative measures appear to be valuable components in an overall acoustic system for burn severity evaluation. [Work supported by NIH.]

FRIDAY MORNING, 20 MAY 1988

GRAND BALLROOM C, 8:15 A.M. TO 12:02 P.M.

## Session YY. Speech Communication VIII: Speech Production

James H. Abbs, Chairman

Waisman Center, University of Wisconsin, Madison, Wisconsin 53706

Chairman's Introduction—8:15

### Contributed Papers

8:17

**YY1. Videostroboscopic images associated with glottographic waveforms.** Gerald S. Berke, Bruce R. Gerratt, and David G. Hanson (VA Medical Center West Los Angeles, W112C, Los Angeles, CA 90073 and UCLA School of Medicine, Los Angeles, CA 90024)

Knowledge of vocal fold vibratory morphology in normal and pathologic phonation is important in understanding glottal airflow and acoustics. Although noninvasive measures of phonation such as photoglottography and electroglottography provide some information on vocal fold movement, inferences are required regarding which glottal events correspond to particular points on the glottographic signals. Thus confirmation of the relationship of glottography to the vibratory movements of the vocal folds is desirable and especially important when studying dysphonia. Here a new technique is reported in which sequentially obtained videostroboscopic images are associated with specific vocal fold vibratory events recorded glottographically. The system involves digitizing glottographic signals simultaneous with the horizontal sweep synch of the video signal and a 5-ms synchronization impulse that is also recorded on the audio channel of a video recorder. The advantages of the system include the ability to sample many images and applicability to clinical investigation. Validation of the technique along with the demonstration of the images and waveforms obtained will be presented. [Work supported by a VA Technical Merit Review Grant.]

8:32

**YY2. Behavior of a dynamic mechanical model of a larynx.** Kevin Brown (Department of Mechanical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139) and Corine Bickley (Room 36-521, Department of Electrical Engineering and Computer Science and Research, Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

A dynamic mechanical model of an adult larynx was constructed. The components that represent the cartilages were machined out of metal. The sizes of the cartilages were taken from measurements reported in the literature [Maue (1970); Kahane (1982)]. A mechanism was designed to allow the arytenoid cartilages to be moved in a manner similar to the complex, natural movements. The vocal folds were simulated using natural rubber. The length of the vocal folds was based on measurements by

Hirano and his colleagues [Hirano *et al.* (1981)]. The model exhibits the relative motions of the laryngeal cartilages and the vibration of the vocal folds. The vocal folds of the model vibrate in a manner similar to observed vibrations of human vocal folds. The fundamental frequency of vibration, or pitch, increases as tension is increased on the vocal folds. Measurements of the fundamental frequency of vibration, the pressures above and below the glottis, and the airflow through the glottis for various conditions of tension, mass, stiffness, glottal area, and profile of the vocal folds of the mechanical model will be reported. The behavior of the model is compared to measurements of pressure and airflow in previously reported static models and interpreted in light of current theories of vocal-fold vibration.

8:47

**YY3. An esophageal electrode for studying posterior cricoarytenoid muscle activity during speech.** Mihoko Fujita, Christy L. Ludlow, Gayle E. Woodson, and Ralph F. Naunton (Speech Pathology Unit, Building 10, Room 5N226, NINCDS, NIH, Bethesda, MD 20892 and UCSD School of Medicine, San Diego, CA 92103)

An esophageal surface electrode was developed for the study of posterior cricoarytenoid (PCA) muscle activity during speech based on an initial design of Woodson and her colleagues. The electrode consists of two multiple strand wires contained in a 5 French pediatric feeding tube. It is passed through one nasal passage into the upper part of the esophagus and picks up PCA activity through the anterior wall of the esophagus. Deep inhalation is a verifying gesture with an absence of activation during sustained modal vowel production. Use was attempted with 19 subjects; 3 normal speakers and 16 patients. The expected activation patterns were obtained in 14 (73.7%) subjects. Problems that interfered with obtaining accurate recordings included a lack of descent into the esophagus and pickup of phonatory vibration. This electrode provides a valuable tool for phonetic and phonatory studies of PCA activation patterns in normal speakers and patients with voice disorders.

9:02

**YY4. The crico-thyroid muscle in voicing control.** Anders Lofqvist, Thomas Baer, Nancy S. McGarr, and Robin Seider Story (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Initiation and maintenance of vibrations of the vocal folds require suitable conditions of adduction, longitudinal tension, and transglottal air flow. This study explores the control of voicing and voicelessness in speech with particular reference to changes in the longitudinal tension of the vocal folds. Electromyographic recordings were made from the crico-thyroid muscle in two speakers of American English and one speaker of Dutch. The linguistic material consisted of reiterant speech made up of CV syllables where the consonants were voiced and voiceless stops, fricatives, and affricates. The results indicate that the crico-thyroid activity increases substantially at the transition from a vowel to a voiceless consonant but not during the transition from a vowel to a voiced consonant. The activity level of the crico-thyroid is thus higher for a voiceless than for a voiced consonant. Measurements of the fundamental frequency at the beginning of a vowel following the consonant show the commonly observed pattern of higher  $F_0$  after voiceless consonants. It is thus suggested that the crico-thyroid muscle shows a different pattern of activation for voiced and voiceless consonants with higher activity for the latter. This finding most likely reflects a difference in the tension of the vocal folds associated with the voicing condition. The same mechanism also seems to contribute to the well-known difference between  $F_0$  at the beginning of vowel following voiced and voiceless consonants.

9:17

**YY5. Fluid mechanics of the velopharyngeal tract.** Ronald C. Scherer (Denver Center for the Performing Arts, 1245 Champa Street, Denver, CO 80204)

Prediction of geometrical characteristics of the velopharyngeal tract (VPT) during speech would aid in determining speech adequacy and intervention effectiveness. Earlier work [beginning with D. W. Warren and A. B. DuBois, *Cleft. Pal. J.* 1, 52-71 (1964)] attempted to predict cross-sectional area of the VPT from measurements of mean flow through the tract and mean pressure drop across the tract by using a discharge coefficient (DC) expression where the DC is constant. In this study, the assumption of a constant DC is challenged through the use of a theoretical equation applicable to similar air channels. The equation takes into account the geometry of the VPT, and includes kinetic pressure terms for the VPT entrance and exit and a viscous pressure term for the duct. The predicted DC significantly increases with Reynolds number (Re) (especially below 1000), asymptotes to higher values for larger VPT cross-sectional area, and is most sensitive to geometry change for low Re. The predicted DC is higher than found in earlier modeling studies. Results suggest that earlier physical models were too simple and further empirical study is justified.

9:32

**YY6. Wisconsin x-ray microbeam system: Articulatory pellet tracking specifications.** Robert D. Nadler, Phillip E. Robl, Daniel J. Wahl, John R. Westbury (University of Wisconsin, Waisman Center, 1500 Highland Avenue, Madison, WI 53706-2280), and Osamu Fujimura (AT&T Bell Laboratories, Murray Hill, NJ 07974)

The x-ray microbeam system for studying tongue movements and other articulatory gestures has been in routine operation for the past year. The precision and accuracy with which the system can track pellets is dependent upon a number of factors. These include the x-ray system operating parameters and characteristics, pellet size, background conditions, and image processing algorithms used for real-time tracking. The results of studies that address these issues, and their importance with respect to interpretation and use of articulatory movement data, will be discussed. Other methodological issues of importance to potential users include head movement correction via the use of reference pellets and empirical specification of anatomical reference planes for expressing the acquired speech movement data. The x-ray dosage incurred by subjects studied to date has also been carefully examined. Even when using peak spatial dose, a conservative estimate, subjects can participate in repeated speech production studies without exceeding the yearly Federal whole body dose limit. The status of the system and planned updates will also be presented. [Support for this facility is provided by NINCDS (NS-16373).]

**YY7. Wisconsin x-ray microbeam system: Subject variables and experimental procedures.** Roxanne DePaul, Robert Nadler, Toyota Ishii, and Carl Johnson (Waisman Center, University of Wisconsin, Madison, WI 53706)

In this first year of operation of the x-ray microbeam laboratory for the study of speech movements, certain procedures and methods have been devised and/or have evolved from practical experience. These include guidelines for the recruitment, screening, and selection of human subjects, along with acquisition of information regarding their anthropometric, linguistic, neurological, educational, dental, and auditory characteristics. Likewise, attempts have been made to devise consistent methods for placement of articulatory pellets as well as for across-subject standardization of those pellet placements. In addition, for pellets on each articulatory site, movement bandwidths and associated sampling rates have been empirically determined. In parallel, methods have been tested and refined for simultaneously sampling speech acoustic and other physiological signals as well as their postacquisition temporal alignment with pellet movement data. Other methods of experimental importance include techniques for cuing subjects and for presentation of utterance material. This presentation will cover these matters with the goal of providing society members with additional x-ray microbeam system familiarity as well as a forum for identification and discussion of yet other methodological issues. [Supported by NIH Grant NS-16373-8.]

10:02

**YY8. Wisconsin x-ray microbeam system: Acquired articulatory data and some preliminary analyses.** James H. Abbs, Robert D. Nadler, Caylee Nychis-Florence, and Nadine P. Connor (Waisman Center, University of Wisconsin, Madison, WI 53706)

To complement the two previous papers, it is useful to provide a more detailed description of data obtained on the new x-ray microbeam system. In the experiment to be described, two-dimensional movements were acquired for pellets placed on the mandible, tongue dorsum, tongue blade, and tongue tip as well as for two reference pellets on a vertical strut attached to an eye glass frame (for head movement corrections). Sampling rates reflected individual movement speeds: 180 samples/s (SPS) for the tongue tip, 90 SPS for the tongue blade, tongue dorsum, and mandible, and 45 SPS for the reference pellets. Four adult normal subjects each produced 50-75 repetitions of the utterances "hot apple," "hat apple," "haddock," and "hassock." Primary analyses focused on the contribution of these four movements to anterior lingual consonant productions, including 2-D pellet positions at palatal contact, and the velocity, amplitude, and direction of the trajectories in achieving those positions. Position and trajectory patterns, along with variability distributions, provide certain perspectives on the motor control of these gestures. [Work supported by NINCDS Grant 16373-8.]

10:17

**YY9. Alterations in lip and jaw kinematics as a function of speech rate.** Nadine P. Connor, James H. Abbs, Caylee Nychis-Florence (University of Wisconsin, Waisman Center, Madison, WI 53706), and Michael P. Caligiuri (VA Medical Center, San Diego, CA 92161)

Manipulation of speech rate provides a means for examining variations in articulatory kinematics with natural alterations in speech motor timing. Such rate studies have been popular, based upon expected insights into underlying speech motor control processes. However, results from prior work on kinematic variations with speech rate have been equivocal. The divergent findings may be due to task differences, the few subjects studied in each experiment, and/or limitations in the number of task repetitions. In this context, each of ten normal subjects produced syllable repetition sequences and sentences with relatively large numbers of repetitions for each. Multiple measures of lip and jaw kinematics were applied, including some previously shown to be particularly revealing in studies of locomotion, arm, and hand movements. Findings indicated previously unreported speech rate-related variations in peak velocity-amplitude ra-

tios, syllable-to-syllable magnitude and timing, vowel-consonant versus consonant-vowel movement durations, and peak velocity time-shifts. The results provide some new perspectives on speech kinematic control. [Work supported by NINCDS Grant 13274-12.]

10:32

**YY10. Labial closing kinematics reconsidered.** Gerald D. Lane and Richard O. DeWald (Department of Linguistics, University of Texas at Austin, Austin, TX 78712)

Ladefoged and Maddieson [UCLA Working Papers 64, 1-137 (1986)] have spoken of "the forceful coming together of the tongue or lips in a stop consonant" as "an attempt to throw one part of the vocal tract through another" (p. 3), while Gracco and Abbs [Exp. Brain Res. 65, 156-166 (1986)], in examining closure into the first /p/ for "sapapple," found peak velocity was attained well before contact. This deceleration into closure, plus "relatively high correlations between the amplitude and peak velocity of individual movements" were taken as evidence that "the labial closure goal was not achieved simply by slamming the lips together" (p. 159). While these descriptions are not necessarily contradictory, they do suggest two different possible movement types. Gracco and Abbs' smooth, centrally peaked velocity profiles resemble those of pointing or reaching. Ladefoged and Maddieson describe, instead, a striking gesture. In order to distinguish these, special care must be taken (1) to ascertain time of contact accurately, (2) not to distort events near impact by excessive low-pass filtering, and (3) to avoid effects of secondary labial articulations. When these precautions are taken, closure into /p/ resembles a blow or tap, with minimal if any deceleration before contact.

10:47

**YY11. Extracting dynamic parameters from articulatory movement.** Richard S. McGowan, Caroline L. Smith, Catherine P. Browman, and Bruce A. Kay (Haskins Labs, 270 Crown Street, New Haven, CT 06511)

In order to model the movements of the speech articulators using a dynamical systems approach, it is necessary to specify (1) the specific form of the dynamical equation, (2) the portion of the articulatory trajectory being described/generated by the equation—the "window," and (3) the values of the coefficients to be used in the equation. In this work, a damped mass-spring dynamical system is assumed, as used in previous task dynamic modeling [Saltzman *et al.*, J. Acoust. Soc. Am. Suppl. 1 82, S15 (1987)]. A program was developed to determine the values of the coefficients for the equations, using various definitions of the articulatory window. The program uses a nonlinear least-squares algorithm to determine the best fit between the observed trajectories and the trajectories generated using a range of values for the coefficients. This program is currently being tested on simulated data with known parameters in order to establish the accuracy and limitations of the procedure. There will be a report on the reliability of the analyses performed using this approach, with an emphasis on numerical considerations. [Work supported by Grant BNS-8520709 from NSF and Grants HD-01994 and NS-13617 from NIH.]

11:02

**YY12. Kinematic characteristics of children's post-vocalic labial stop consonants.** Bruce L. Smith and Ann McLean-Muse (Department of Communication Sciences and Disorders, Northwestern University, Evanston, IL 60208)

In English, vowels tend to be longer preceding voiced versus voiceless obstruents. Although perceptual factors are often thought to be primarily responsible for this effect, part of the difference may also be a result of inherent speech production factors. Adults commonly move their articu-

lators more rapidly when closing into voiceless versus voiced obstruents, which may contribute to shorter vowels preceding voiceless obstruents. Despite several acoustic studies of this general phenomenon in English-learning children, however, little developmental information is available concerning articulatory characteristics of children's post-vocalic consonants. Although children show acoustic vowel duration differences in these contexts, it is not known whether articulatory differences occur in their productions, as they do for adults. One purpose of this study was to obtain kinematic data from children concerning this issue. Articulatory peak velocity data for upper lip, lower lip, and jaw movements were obtained for eight children in each of three age groups and eight adults. A peak velocity difference for [p] vs [b] closing gestures was observed for each of the four age groups. There was also a tendency for the difference between [p] and [b] peak closing velocity to increase with age.

11:17

**YY13. Native speakers of Spanish and English compensate equally well for a bite block in producing vowels.** J. E. Flege (Biocommunication, University of Alabama at Birmingham, University Station, Birmingham, AL 35294)

A recent detailed study [Flege *et al.*, J. Acoust. Soc. Am. 83, 212-228 (1988)] provided auditory and acoustic evidence that talkers do not compensate completely or instantly when producing vowels with a bite block. This study examined compensation physiologically. Adult native speakers of English and Spanish produced nonsense disyllables (/bVba/) in a Spanish or English carrier phrase. Glossometry was used to measure the vertical distance of the tongue from the hard palate at four sensor locations in multiple productions of the stressed vowels /i/, /e/, /a/, /o/, and /u/. Even though Spanish has fewer vowel categories than English (5 vs 15), the Spanish and English subjects showed equal variability in tongue positions (standard deviations of 0.746 and 0.756 mm). The subjects in both groups showed significantly greater variability in the bite-block than normal-speech condition (means s.d.'s of 0.821 vs 0.722 mm). There were clear differences in the average tongue positions for Spanish and English vowels spoken normally. The eight subjects in both groups showed tongue position differences (unsigned) of only about 1 mm when producing vowels normally and with a bite block. The effect of condition on tongue positions was not significant for either group. [Work supported by NIH.]

11:32

**YY14. Intonational and rhythmic correlates of stress clash.** Mary Beckman, Kenneth deJong (Department of Linguistics, 204 Cunz Hall, Ohio State University, Columbus, OH 43210), and Jan Edwards (Hunter College of Health Sciences, CUNY, New York, NY 10021)

Phonologists support a directly metrical representation of stress rhythms by observing that when one stress "beat" follows closely on another, the rhythmic clash seems to be remedied either by retracting the first stress, as in *Chinese exports*, or by increasing the interstress interval, so that the *pop of pop posed* is longer than that of *poppa posed*. Cooper and Eady [J. Mem. Lang. 25, 369-384 (1986)] have questioned the empirical validity of such observations by showing that the first syllable's duration is not longer in phrases where stress should have retracted to it; nor is it longer in *Chris Mack* compared to *Chris MacKensie*. Recent experiments, however, support the notion of rhythmic clash even if the precise formulation of metrical phonologists is wrong. It was found (1) that syllables perceived as bearing a retracted stress have longer durations relative to the less stressed following syllable, (2) that phrases with retracted stress have different distributions of pitch accents in the intonation pattern, and (3) that although the *pop of pop posed* is not longer than that of *pop opposed*, it is longer than that of *poppa posed*. These results are used to support an indirect metrical representation in which stress patterns are read from a prosodic constituent tree that includes intonational structure.

**YY15. Phonological constructs: Electromyographically obtained speech error evidence.** Richard A. Mowrey (594 Besserer Street, Ottawa, Ontario K1N 6C9, Canada) and Ian R. A. MacKay (Phonetics Laboratory, Linguistics Department, University of Ottawa, Ottawa, Ontario K1N 6N5, Canada)

In recent years, speech error data have often been cited as evidence for the existence of, or for the psychological reality of, certain phonological constructs, such as the phoneme or the distinctive feature. Further claims have been made concerning the role of these units in the processing of

motor control instructions in the production of speech output. This work reports on the collection of phonological speech error data in a controlled laboratory environment using Basmajian electrode EMG in various speech muscles, as well as audio recordings. Both integrated EMG signals and single motor unit analysis are presented. The results lead to the conclusion that published speech error corpora based on field-collected errors are skewed by the combined problems of collection technique, categorical perception, and hearer normalization, and that this skewing leads to circularity in the arguments based on these corpora. This evidence provides support for individual motor instructions as phonological primitives in motor speech control, and does not support higher-order motor control units such as features and segments.

FRIDAY MORNING, 20 MAY 1988

GRAND BALLROOM B, 8:30 TO 11:35 A.M.

## Session ZZ. Psychological Acoustics VI, Physiological IV, and Noise VI: Effect of Noise on the Auditory System

Julia D. Royster, Chairman

*Environmental Noise Consultants, Inc., P. O. Box 144, Cary, North Carolina 27512-0144*

Chairman's Introduction—8:30

### Contributed Papers

8:35

**ZZ1. Objective measures of the annoyance due to automobile passenger compartment noise.** John S. Lamancusa (Mechanical Engineering Department, Penn State University, University Park, PA 16802)

The annoyance attributable to complex noises consisting of multiple pure tones (which may or may not be harmonically related) and substantial broadband noise is not easily quantifiable. It has been shown that dBA is a poor indicator of loudness [Hellman and Zwicker, *J. Acoust. Soc. Am.* **82**, 1700-1705 (1987)], and a number of other measures have been proposed. In this study, the results of subjective preference tests of passenger compartment noise are compared to objective measures of loudness and annoyance. Eight synthesized variations of the noise experienced at full throttle acceleration were ranked by 40 subjects according to their desirability. All samples were normalized to 101-dB linear. These results are compared to the objective rating methods: AI (articulation index), ISO 532B (Zwicker method), Stevens Mark VII, CRP (Composite Rating of Preference), and PNL (perceived noise level). The CRP offers the best correlation with subjective results. The results of Mark VII, and ISO 532B are markedly improved by the addition of empirically derived corrections.

8:50

**ZZ2. Advanced turboprop aircraft flyover noise: Annoyance to counter-rotating-propeller configurations.** David A. McCurdy (NASA Langley Research Center, Mail Stop 463, Hampton, VA 23665-5225)

Two experiments were conducted to quantify the annoyance of people to the flyover noise of advanced turboprop aircraft with counter-rotating propellers (CRP). The objectives were: (1) determine the effects of tonal content on annoyance; and (2) compare annoyance to advanced turboprop aircraft with annoyance to conventional turboprop and jet aircraft. A computer system was used to synthesize realistic, time-varying simulations of advanced turboprop takeoff noise. For the first and second experiments, respectively, the system generated 27 noises representing CRP configurations with an equal number of blades on each rotor, and 35

noises representing CRP configurations with an unequal number of blades on each rotor. Included in each experiment were five conventional turboprop and five conventional jet takeoffs. Each noise was presented at three levels to subjects who judged the annoyance of each stimulus. Analyses of the judgments examine the effects on annoyance of blade passage frequency, tone-to-broadband noise ratio, and aircraft type. The annoyance prediction ability of current noise metrics is also examined.

9:05

**ZZ3. Quality assurance in automated monitoring of high-energy impulsive sounds.** George A. Luz, William A. Russell, and Nelson D. Lewis (U.S. Army Environmental Hygiene Agency, Aberdeen Proving Ground, MD 21010-5425)

U.S. Army policy is to measure noise levels whenever the projections of noise contouring computer programs show a noise environment "unacceptable" for residential use extending beyond the Federal installation's boundary. The threshold for "unacceptable" is an A-weighted day-night level (DNL) of 75 dB for aircraft and a C-weighted DNL of 70 dB for high-energy impulsive sounds. For adequate quality assurance (QA) in monitoring high-energy impulsive sounds, statistical decision criteria are employed including threshold, duration, number of events per unit time, peak-to-SEL differences, and event probability distributions. The strengths and weaknesses of each QA criterion is discussed.

9:20

**ZZ4. Annoyance factors for common neighborhood noise.** Lynn S. Alvord (Department of Communication Disorders, 1201 Behavioral Science Building, University of Utah, Salt Lake City, UT 84112)

Neighborhood noise is treated separately from aircraft or traffic noise in most local noise ordinances. Annoyance factors for common neighborhood noise were rated by 63 subjects in Salt Lake City, Utah. Subjects were asked to rate on a scale of 1 to 5, the degree to which each factor contributed to total annoyance. Most frequently mentioned annoying

noise sources were dogs (38.1%), followed by sirens (12.7%). The following sources were also identified at higher than 3% incidence: garbage trucks, buses, children playing, doors slamming, noisy neighbors, helicopters, and metal stairways. Highest-rated annoyance factors were: loudness, time of occurrence, frequency of occurrence, sound quality, and interference with sleep. Relative importance of annoyance factors found in this study differ from those of previous studies, which dealt more with aircraft or traffic noise. Fear and interference with conversation and sleep are of less importance for "neighborhood" noise than aircraft and traffic noise [T. J. Schultz, *J. Acoust. Soc. Am.* **64**, 377-405 (1978)]. For neighborhood noise "meaning" the sound portrays is an important factor in certain situations. The importance of "situational factors" found here emphasizes the need for increased use of specific prohibitions in local noise ordinances.

9:35

**ZZ5. An overview of occupational noise regulations in Canada.** Deirdre A. Benwell (Bureau of Radiation and Medical Devices, National Health and Welfare, Room 233, EHC, Ottawa K1A 0L2, Canada)

In Canada, there are occupational noise regulations at both provincial and federal levels of government. An overview of present Canadian occupational noise legislation is given. In addition, recent Canadian activities concerning occupational noise exposure standards, guidelines, and other important background documents are described. The various methods used to assess compensation for occupational hearing loss are also summarized. Some recommendations are made for future activities in this area.

9:50-10:05

Break

10:05

**ZZ6. The critical exposure in acoustic trauma.** W. Dixon Ward (Hearing Research Laboratory, University of Minnesota, 2630 University Avenue S.E., Minneapolis, MN 55414)

As one increases the intensity  $I$  of a noise exposure of fixed duration  $t$ , the resulting damage grows gradually up to a critical point at which the damage increases precipitously. The intensity at which this occurs, however, is not independent of the duration, so it cannot be called a "critical intensity." Neither is this point dependent only on the total energy of the exposure  $It$ : Less energy is required at shorter durations in order to reach the critical point, so even "critical energy" would be a misnomer. Instead, the implication of an extended series of experiments in which chinchillas were given single exposures of a few minutes to 1.5 days at levels up to 120 dB SPL is that the critical exposure appears to be defined by  $I^2t = C$ , or equivalently,  $p^2t = k$ , where  $k$  is about  $10^6 \text{ Pa}^4 \text{ s}$  for the chinchilla. This value is in agreement with recent results defining critical exposures to impulse noise in the chinchilla reported by Patterson and Hamernik [*J. Acoust. Soc. Am.* **81**, 1118-1129 (1987)]. The meager data in the literature suggest that for man,  $k$  may be around  $10^{10} \text{ Pa}^4 \text{ s}$ . [Work supported by NINCDS Grant 12125.]

10:20

**ZZ7. Influences of temporal spacing of impulse noise on temporary and permanent threshold shifts.** Richard W. Danielson (Callier Center, University of Texas at Dallas, Dallas TX 75235), Donald Henderson, and Samuel S. Saunders (Department of Communication Disorders and Sciences, State University of New York at Buffalo, Buffalo, NY 14266)

Chinchillas were exposed to impulse noise conditions with equal acoustic energy and spectra but with three different temporal patterns and two different amplitude levels (135 and 150 dB SPL, counterbalanced by 15-dB trade-offs in number of events). Auditory-evoked responses were used to measure TTS and PTS. Degree of hearing loss and pattern of recovery were not uniform among the groups, contrary to assumptions related to the equal energy hypothesis. Relative to hearing loss after exposure to one impulse/second: (1) less hearing loss was seen after exposures to bursts of 1 impulse/50 ms and (2) greater hearing loss was seen when high-level impulses occurred in salvos (pairs of impulses presented at 1-s intervals, with 50-ms interpulse interval in each pair). Findings will be compared to reports of a possible "period of vulnerability," during which hearing loss may be more extreme if the ear is exposed to additional impulse noise while the ear is still recovering. [Work supported by NIOSH and the Department of the Army.]

10:35

**ZZ8. Altered susceptibility of the inner ear following repeated noise exposure.** Daniel J. Franklin, Barden B. Stagner, Brenda L. Lonsbury-Martin, and Glen K. Martin (Department of Otorhinolaryngology and Communicative Sciences, Baylor College of Medicine, Houston, TX 77030)

The susceptibility of the inner ear to repeated noise exposure was investigated in the rabbit using tests of behavioral thresholds and distortion-product emissions (DPEs). Behavioral thresholds, measured by a classical-conditioning technique, were collected at 11 frequencies representative of the rabbit hearing range. Acoustic distortion products at the  $2f_1 - f_2$  frequency were measured as both DPE audiograms, generated by equilevel primaries at 45, 55, and 65 dB SPL, and input-output functions in 5-dB steps at nine frequencies. Following acquisition of control measures of DPEs and behavioral thresholds, DPEs were analyzed at regular intervals during both exposure and recovery periods and compared to corresponding behavioral measures. In four animals, following repeated episodes of exposure (octave band of noise centered at 1 kHz at 95 dB SPL) and recovery (3 weeks), both DPE amplitudes and behavioral audiograms revealed an increasing resistance to the effects of noise exposure. Comparison with the amount of amplitude reduction in a 4-kHz DP caused by a 5-min exposure during the preexposure periods to a 95-dB SPL tone at 4.215 kHz (1/2 octave below the geometric mean of the primaries) showed a poor correlation with the lessened susceptibility to repeated noise exposure. The concept describing a dynamic mechanism of repair of sensory elements proposed by a number of investigators to explain recovery from sound exposure may account for the increasing resistance to the effects of repeated noise exposure as detected by both DPEs and behavioral responses. [Work supported by NINCDS.]

10:50

**ZZ9. Intracochlear pressures resulting from brief high intensity stimuli.** James H. Patterson, Jr., Ben T. Mozo (U.S. Army Aeromedical Research Laboratory, P.O. Box 577, Fort Rucker, AL 36362-5292), Karl Buck, and Laurent Decory (Franco-German Institute of Research, Saint-Louis, France)

A previous study has shown similar hearing loss and cochlear injury from exposures to impulses differing in peak pressure, but having the same energy and Fourier pressure spectrum. Nonlinear processes in the middle ear or transmission through the external and middle ear may render different stimuli more similar once they reach the cochlea, thus accounting for the equivalent injury observed. In order to explore this possibility, pressures in the basal turn of the cochleas of eight anesthetized chinchillas



were measured for impulses presented at levels between 88-and 147-dB peak. This included the impulses used in the previous injury study. While nonlinearities were noted at the higher intensities, there was a remarkable preservation of the pressure-time histories. In addition to these impulse stimuli, intracochlear pressures were measured using brief tone pips whose envelope was Gaussian. Frequencies of 700, 1375, 2000, 2700, and 4100 Hz were chosen to span the frequency response of the speaker. With these stimuli, nonlinearities appeared as both harmonic and subharmonic distortion as the intensities increased.

11:05

**ZZ10. Changes in frequency selectivity in the chinchilla following a noise induced permanent threshold shift.** Robert I. Davis, William A. Ahroon, and Roger P. Hamernik (Auditory Research Laboratory, SUNY at Plattsburgh, Plattsburgh, NY 12901)

Evoked-potential tuning curves (TC) were obtained on 106 chinchillas before and after acoustic overstimulation in order to assess the effects of the magnitude of hearing loss on frequency selectivity. Pre- and postexposure measures of auditory thresholds and masked thresholds (simultaneous tone-on-tone paradigm) were obtained in each animal at 0.5, 1, 2, 4, 8, and 11.2 kHz, using the evoked auditory response recorded from the inferior colliculus. Three TC variables ( $Q$ -10 dB, tail-tip difference, and the high-frequency slope) and sensory cell losses were compared to the amount of noise-induced permanent threshold shift (PTS) produced by a variety of noise exposures. Based upon large sample averages, frequencies showing  $PTS > 20$  dB also showed statistically significant differences between pre- and postexposure measures of all three TC variables. For  $10 < PTS < 20$  dB only the tail-tip difference showed a statistically significant change, while for  $PTS < 10$  dB there were no measurable changes in

the TC variables. The percentage of outer hair cell loss showed an orderly and systematic increase as PTS increased and as TC variables changed across the entire range of test frequencies. The inner hair cells were essentially unaffected. These results show that there is a systematic change in the TC variables that define the quality of tuning as hearing loss progressively increases and that these changes are clearly related to outer hair cell losses. [Research supported by NIOSH and DOD.]

11:20

**ZZ11. "Enhanced" evoked response amplitudes in the chinchilla following acoustic trauma.** R. Salvi, S. Saunders, N. Powers, and M. A. Gratton (Department of Communication Disorders and Sciences, 109 Park Hall, SUNY University of Buffalo, Buffalo, NY 14214)

The amplitudes of auditory-evoked responses recorded from chronic electrodes in the inferior colliculus of the chinchilla were measured before and after acoustic trauma. Acoustic trauma was induced using a 2-kHz continuous tone that resulted in either 30-40 dB of TTS or PTS between 2-8 kHz. The high-intensity exposures resulted in systematic changes in the input/output functions of the evoked response. The most striking change was an increase in the maximum amplitude ("enhanced") of the evoked response at frequencies below and at the low-frequency edge of the hearing loss (0.5 and 2 kHz). By contrast, the maximum amplitude seen at frequencies near the middle of the hearing loss or its high-frequency border (4 and 8 kHz) was generally depressed. In addition to the change in maximum amplitude, there were also changes in the slope of the evoked response input/output functions. The results will be related to the pattern of hair cell loss as well as to possible underlying neural mechanisms. [Work supported by NIH R01 NS16761 and NS23894.]

FRIDAY MORNING, 20 MAY 1988

WEST BALLROOM B, 8:30 A.M. TO 12:05 P.M.

### Session AAA. Underwater Acoustics VIII: Underwater Propagation

Kenneth E. Gilbert, Chairman

*The National Center for Physical Acoustics, University of Mississippi, University, Mississippi 38677*

Chairman's Introduction—8:30

#### *Contributed Papers*

8:35

**AAA1. PARQUES: A range-dependent environment propagation loss model.** Bruno Grandvilllemin (DCAN Toulon-GERDSM, Six-Fours les Plages 83140, France)

A proposal to solve the Helmholtz's equation by means of the parabolic approximation in a two-dimensional range-dependent environment will be given. It will make clear the solution for a quadratic development of the propagation operator that allows the consideration of a larger angular aperture of the source. The numerical solution is given by an algorithm of finite difference using the Crank-Nicholson method. The problem of rectilinear, horizontal, and oblique interfaces between two fluid media is solved. The problem of a range variable sound velocity profile is treated by conditions of continuity between two media with the given profiles. The regressive wave is also computed. The results of numerical simulations executed on the array processor FPS 164 are presented. Also are shown comparisons between theoretical computation and experimental data for range-independent and range-dependent environment.

8:50

**AAA2. Adaptation of a three-dimensional ray tracing code for use with gridded input oceanographic models.** J. Dougherty, A. E. Newhall, J. F. Lynch (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543), and C. S. Chiu (Institute for Naval Oceanography, NSTL, MS 39529)

In order to be able to adapt the Hamiltonian ray tracing program (HARPO) developed by NOAA's Wave Propagation Laboratory for use with gridded numerical oceanographic models, three different approaches have been employed. The first method employs using a 3-D Fourier decomposition of the sound-speed field, giving one the ability to use fast Fourier techniques and spatially filter input data but being limited to a constant grid with mesh determined by the highest spatial frequency along a given coordinate. The second technique employs an empirical orthogonal function analysis of the sound-speed field, the coefficients of which are splined to ensure the continuity requirements of the ray trace. This method allows one to: (1) simply filter the field by mode energy; (2) project easily on oceanographic modes; and (3) vary the sampling to meet

the local ocean conditions, thus saving on computer memory requirements. The final technique employed is "feature hanging," which combines using analytic forms for ocean features and gridded input for their positions. This is an extension of the existing HARPO feature to be consistent with current oceanographic models. Example calculations using each technique will be presented.

9:05

**AAA3. The time-marched FFP for modeling ocean acoustic pulse propagation.** Michael B. Porter (SACLANT Undersea Research Centre, APO, NY 09019)

Traditional ocean acoustic models for cw (time-harmonic) signals may be extended in an obvious fashion for broadband problems using Fourier synthesis. Alternatively, many of these models may also be solved directly in the time domain that raises the questions of what is the optimal solution and under what conditions. A time-marched fast-field program has been developed in which the solution is represented as a sum of Fourier components in range with time and depth-dependent amplitudes,  $a(z,t,k)$ . These amplitudes each satisfy a simple hyperbolic equation that is discretized in depth using finite elements and in time by a simple explicit integrator. Snapshots of the ocean acoustic field are then obtained in the usual fashion using an FFT to assemble the various spatial Fourier components. Examples of the method applied to pulse propagation in certain ocean acoustic scenarios will be presented with particular regard to the question of the strengths and weaknesses of the time-marched FFP versus synthesis from time-harmonic FFP solutions.

9:20

**AAA4. Intrinsic modes revisited.** F. Xiang, M. Cada (Department of Electrical Engineering, Technical University of Nova Scotia, Halifax, Nova Scotia B3J 2X4, Canada), and L. B. Felsen (Department of Electrical Engineering and Computer Science, Polytechnic University, Farmingdale, NY 11735)

Intrinsic modes [J. M. Arnold and L. B. Felsen, *J. Acoust. Soc. Am.* **76**, 850–860 (1984)] are wave objects that propagate without coupling upslope or downslope in a homogeneous ocean wedge above a penetrable bottom. They have been observed experimentally and have been referred to as wedge modes in that context [F. B. Jensen and C. T. Tindle, *J. Acoust. Soc. Am.* **82**, 211–216 (1987)]. The exact analytic form of the intrinsic modes (IM) involves a spectral integral which, for small slopes, can be reduced asymptotically to a simpler integral expression that leads to the conventional adiabatic modes where these apply but is valid also through the upslope cutoff transition. Except for small slopes, this simplified form upslope or downslope is flawed by energy balance defects beyond upslope cutoff and by inaccuracies of the field leaked into the bottom. The exact form of the IM has now been reexamined and restructured so as to be computable directly. In fact, the convergence of this restructured form, which tracks multiple reflected plane wave spectra, improves with slope angle, thereby making it complementary to the small slope approximated algorithm. Calculation of the complete IM removes the defects noted above and thereby establishes the IM firmly as the analytical wave object that describes the experimentally observed wedge modes. Numerical comparisons of the IM with wedge solutions obtained by the parabolic equation and coupled mode codes confirm this conclusion. [Work supported by ONR Ocean Acoustics.]

9:35

**AAA5. An integral equation approach to range-dependent boundary value problems in ocean acoustics.** Peter H. Dahl (MIT/WHOI Joint Program in Oceanography/Oceanographic Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543) and George V. Frisk (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

In ocean acoustics, the introduction of range discontinuities, for example, the water-to-ice canopy surface, creates a mixed boundary value problem. In this paper, an exact solution of certain mixed boundary value problems is discussed using the Wiener-Hopf method. A key attribute of this approach is that it is not fundamentally numerical in nature and allows additional insight into the mathematical and physical structure of the acoustic field due to range discontinuities. The problem discussed here is a canonical one: A plane wave is incident upon a planar surface where the boundary condition changes from Dirichlet (free surface) to Neumann (hard). The boundary conditions addressed in this problem are highly idealized renditions of what happens, when, say, a plane wave is incident upon a water-to-ice canopy surface; nevertheless, important features of the diffraction process are produced here, and the solution gives considerable insight into the process. The solution of the diffracted potential is partitioned into two components: a field consisting of cylindrical waves weighted by a polar gain function, resulting from the artificial source created by the discontinuity in the boundary; and a field containing a residue contribution which restores field continuity along the line corresponding to specular reflection. Contour plots of equal pressure amplitude show how the component fields superimpose such that the boundary conditions are maintained, and how energy is redistributed across the angular spectrum in the diffraction process. The latter is related to mode coupling due to boundary changes in waveguide problems. [Work supported by ONR.]

9:50

**AAA6. Gaussian beams and salt tables.** H. P. Bucker (Code 541, Naval Ocean Systems Center, San Diego, CA 92152)

In the study of many ocean acoustic systems it is desirable to perform very rapid calculations of the sound field. For most problems of practical interest, 2-D models provide accurate results. That is, the sound propagation is different along different radials but the interchange of energy between radials can be neglected. In this case, it is possible to use salt tables where sound angle/level and time is tabulated as a function of range. This paper examines the compatibility of the Gaussian beam propagation model and salt tables. Special items covered are the partition of beams between tables, receivers on the surface and bottom boundaries, and the effect of beam displacement associated with boundary reflection. [Work partly supported by the NORDA NOP Program.]

10:05

**AAA7. A space-time split-step algorithm for pulse propagation in refracting media.** Johanan L. Codona (AT&T Bell Laboratories, Whippany, NJ 07981)

A simple, time-domain split-step algorithm for pulse propagation in a refracting medium is derived from the time-dependent wave equation (TDWE). The derivation is similar in form and spirit to the factored Helmholtz equation methods used in finding generalized parabolic equations. Like the Helmholtz equation analyses, the factored TDWE involves nonlocal pseudodifferential operators and a commutator term. Making use of causality, and considering media in which the index-of-refraction changes only slowly over a typical acoustic wavelength, the commutator term can be neglected and the TDWE reduces to a Schrodinger-like equation in space-time. From here, a variety of methods (for example, phase-space path integrals) can be employed to generate marching algorithms for computing the evolution of an acoustic pulse in a refracting medium. A fast split-step algorithm (of the form FFT/multiply/FFT/multiply/etc.) only results if the operator playing the role of the Hamiltonian in the pulse Schrodinger equation is separable, which is only a good approximation for narrow-band, forward propagation. However, a somewhat slower phase-space algorithm based directly on the phase-space path integral does not suffer any of these limitations. The time-domain split-step algorithm will be compared with other pulse-propagation methods (such as direct Fourier synthesis) and the resulting accuracy and numerical efficiencies discussed.

10:20

**AAA8. Computing ocean pulse propagation using parabolic equations and numerical phase correction.** Robert P. Gilbert (Department of Mathematical Sciences, University of Delaware, Newark, DE 19716) and David H. Wood (Code 3332, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

When one uses a parabolic equation instead of the Helmholtz equation to synthesize a pulse propagating in the ocean, the so-called "phase errors" in solutions of the parabolic equation lead to errors in the shape and arrival time of the pulse. Of course, different kinds of parabolic equations are known to have different types and sizes of phase errors. A transmutation approach is used, that is, the kernel of an integral transformation is sought that maps solutions of the parabolic equation into solutions of the Helmholtz equation. Naturally, different types of parabolic equations require different transmutation kernels. Using the appropriate transmutation kernel, the effects of parabolic phase errors can be numerically corrected for. In an ocean with slow horizontal variation, it is possible to periodically update the transmutation kernel as the solution is tracked horizontally. Numerical examples will be presented.

10:35

**AAA9. Computing ideal starting fields for parabolic equations: Approach and examples.** E. Richard Robinson and David H. Wood (Code 3122, New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

A parabolic equation model requires a starting field, that is, the values of the field must be given as a function of depth at a fixed range. The ideal starting field is a weighted combination of propagating normal modes. Since it is time consuming to compute normal modes, the usual practice is to use a more easily computed result such as a Gaussian, filtered Gaussian, sinc, or uniform ocean starting field. However, in our approach, less effort is required because the individual normal modes are not computed but only the desired combination of normal modes that gives the ideal starting field. Two example normal mode starting fields are computed and are compared to other starting fields. The effects of the various starting fields on the propagation loss predicted by a parabolic equation model are shown.

10:50

**AAA10. A wide-angle initial field for parabolic equation models.** David J. Thomson (Defense Research Establishment Pacific, FMO Victoria, British Columbia V0S 1B0, Canada) and C. Sean Bohun (Department of Physics, University of Victoria, Victoria, British Columbia V8W 2Y2, Canada)

Numerical predictions of underwater sound propagation are routinely carried out by applying marching algorithms to solve parabolic equations. These equations require initial field data to begin marching process. For many applications, a suitably normalized Gaussian or sinc function of depth ( $z$ ) is used [E. R. Robinson and D. H. Wood, *J. Acoust. Soc. Am. Suppl.* 1 **81**, S10 (1987)]. These functions, however, do not model correctly the vertical-wavenumber ( $k_z$ ) spectrum associated with the farfield of a point source. In this paper, a new starting field is presented that matches the proper  $k_z$  variation over the full spectral aperture needed by wide-angle parabolic equation models. The new field in the  $z$  domain is obtained numerically via a discrete Fourier transform of its bandlimited spectrum. Because of this efficient distribution of energy in the  $k_z$  domain, a larger step-size ( $\Delta z$ ) can be used than that required by the Gaussian initial field [H. M. Garon, J. S. Hanna, and P. V. Rost, *J. Acoust. Soc. Am. Suppl.* 1 **61**, S12 (1977)]. Numerical examples comparing the effects of different starting fields on wide-angle parabolic equation predictions are presented.

11:05

**AAA11. Phase space and path integral methods in computational acoustics.** Louis Fishman (Department of Mathematics, Colorado School of Mines, Golden, CO 80401)

The development and application of microscopic phase space methods and global path integral constructions to both direct and inverse problems in acoustics are reviewed. The Helmholtz equation is considered for (1) one-way direct wave propagation, (2) two-way direct wave propagation, and (3) inverse formulations. The one-way theory concentrates on the construction of exact, numerical, and uniform perturbation solutions to the Weyl composition equation and their application to the further development and testing of the phase space marching algorithm. Specific comparisons will be made with the IFD codes for one-way wave equations derived from formal operator rational approximations. The two-way theory considers the evaluation of the Feynman/Garrod approximate path integral, the construction of an explicit path integral representation for the homogeneous Helmholtz equation, and the combination of transmutation and quantum field theoretic variable-dimensionality ideas to incorporate backscatter. The inverse analysis seeks to exploit the Fourier integral operator structure of both the finite and infinitesimal propagators and the semigroup property of the one-way Helmholtz equation to exactly solve the refractive index profile reconstruction problem for transversely inhomogeneous environments. [Work supported by NSF, AFOSR, ONR.]

11:20

**AAA12. The effects of dispersion, attenuation, range, and other factors on the duration of transients.** Donald R. Del Balzo, Robert L. Field, and Michael D. Collins (Naval Ocean Research and Development Activity, NSTL, MS 39529)

In this paper, the results of a theoretical investigation into the effects of various factors including dispersion, attenuation, and range on the duration of transients in shallow water ocean environments are presented. An acoustic source with the time history  $s(t;\omega) = 0$  for  $t < 0$  and  $\sin(\omega t)$  for  $t > 0$  is placed in the ocean several kilometers from a receiver. The duration of the transient is the amount of time between the first arrival and the achievement of a time harmonic state at the receiver. Using a range-dependent time domain model developed at NORDA, the Klein-Gordon equation is solved to obtain the received signal  $P(t;\nu)$  due to the Gaussian source  $S(t;\nu) = \exp[-(\nu t)^2]$ . Then,  $P(t;\nu)$  is convolved to obtain the response  $p(t;\omega)$  due to  $s(t;\omega)$ . Qualitatively, one would expect transient duration to increase with dispersion and range and to decrease with attenuation. A presentation of quantitative results to illustrate these relationships and discuss the possibility of the use of transient duration to solve inverse problems will be given. [Work supported by ONR and NORDA.]

11:35

**AAA13. Transient acoustic wave propagation in a fluid layer with stepwise range dependent perfectly reflecting boundaries.** Gary H. Brooke, Roderick E. Glover (Defence Research Establishment Pacific, FMO Victoria, British Columbia V0S 1B0, Canada), and Jacques R. Chamuel (Sonoquest/Advanced Ultrasonic Research, P. O. Box 153, Wellesley Hills, MA 02181)

The coupled mode solution technique [R. B. Evans, *J. Acoust. Soc. Am.* **74**, 188-195 (1983)] is adapted to the problem of transient acoustic wave propagation in a single fluid layer with perfectly reflecting boundaries. Properties of one of the boundaries are allowed to vary in a stepwise range dependent manner. The transient response due to an impulsive line source is synthesized from solutions to a global matrix formulation of the coupled mode problem for two-dimensional waveguides. Several numerical examples are presented that model: (i) waveguide scattering behavior of both single and multiple scatterers placed near the boundary of the fluid layer and (ii) propagation in a waveguide with a variable impedance boundary comprised of alternate free and rigid strips. The numerical results are compared with experimental observations taken from measurements performed at ultrasonic frequencies using an air-suspended waveguide [J. R. Chamuel and G. H. Brooke, *J. Acoust. Soc. Am. Suppl.* 1 **77**, S13 (1985)].

**AAA14. *In situ* acoustic measurements for tactical ASW applications.** Charles H. Wiseman (Lockheed Missiles & Space Company, Inc., Astronautics Division—Advanced Marine Systems, P. O. Box 3504, Sunnyvale, CA 94088-3504)

Temperature versus depth profiles provided by expendable bathythermograph (XBT), together with information from atlases of seasonal water conditions and ocean bottom mapping, constitute the inputs for predicting sound propagation conditions for operational ASW use. At sea, XBT inputs feed computer models and nomograms that furnish inference as to sound propagation conditions on which the ASW tactics can be based—for example, the routing of convoys or naval task units to minimize the likelihood of detection by enemy submarines; the selection of optimum modes for active and passive sonars; or the identification of

targets for over-the-horizon long range weapons systems. While acoustic predictions based on XBT inputs are useful, they can be improved by supplementing them with *in situ* acoustic measurements at sea. The inferences as to sound propagation conditions based only on XBTs are limited by their number and placement considered against the backdrop of inhomogeneities of the ocean. On the other hand, the *in situ* measurement of sound is not an *inference* of sound propagation—in point of fact, it is the direct agent, sound itself. The ASW forces should steal a leaf from the Radio Communicator's notebook. While predictions of hf radio propagation are set forth in special atlases, *in situ* measurements of radio propagation conditions in the form of ionospheric soundings (ionograms) furnish the most accurate, real-time indications of usable frequencies with minimum multipath distortions. There are a variety of methods, overt and covert, which can be used by ships and aircraft to effect *in situ* acoustic measurement and these are discussed in the body of the paper.

FRIDAY MORNING, 20 MAY 1988

ASPEN ROOM, 8:45 TO 11:35 A.M.

### Session BBB. Musical Acoustics V: General Topics in Musical Acoustics

Uwe J. Hansen, Chairman

Department of Physics, Indiana State University, Terre Haute, Indiana 47809

Chairman's Introduction—8:45

#### Contributed Papers

8:50

**BBB1. Dependence of vibrational modes of a handbell on thickness and added mass.** H. John Sathoff, Jiang Zhiqing, and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The vibrational modes of a large tuned handbell ( $G_2$ ) are compared with those of an untuned bell casting having the same shape but with twice the thickness. Doubling the wall thickness is found to raise the frequencies of the various modes by factors ranging from 1.4 to 2.4 (flat plate theory predicts a factor of 2 for all modes). Unlike the tuned  $G_2$  handbell, whose pitch coincides with the frequency of the fundamental mode of vibration, the thick bell has two prominent strike notes, depending upon where it is struck. These subjective tones appear to be created by groups of three nearly harmonic partials radiated by the  $(m, l)$  and  $(m, \#)$  families of modes, respectively. The acoustical effects of attacking up to twelve 35-g brass cylinders have also been observed, both in a symmetrical pattern and in clusters similar to the clusters of *mei* found on ancient Chinese bells [Rossing *et al.*, J. Acoust. Soc. Am. Suppl. 1 77, S102 (1985)]. The additional masses lower the frequencies of certain modes but have relatively little effect on modal decay rates.

9:05

**BBB2. Sound radiation and frequency response of classical and folk guitars.** Eric T. Watson and Thomas D. Rossing (Northern Illinois University, Department of Physics, DeKalb, IL 60115)

The sound radiation has been measured as a function of angle in an anechoic room when the guitar is driven with a sinusoidal force. Classical guitars radiate more strongly than folk guitars when the same force is applied to the bridge [J. Popp and T. D. Rossing, International Symposium on Musical Acoustics, Hartford, July 1986]. Radiation constants are large as 2.7 Pa/N are observed at guitar resonances. Near certain resonances, dipolar or quadrupolar radiation predominates. The radiated

sound field is compared to the calculated and measured force on the bridge when the string is plucked.

9:20

**BBB3. On the structural-acoustic design of guitars.** Evan B. Davis (3517 N. E. 6th Street, Renton, WA 98056)

A structural-acoustic model of a guitar top plate backed by a rigid cavity has been built. Acoustic coupling between the interior and exterior acoustic fields is accomplished with a simple model of the soundhole. Acoustic power spectra are calculated in 1/12 octave intervals from 55 to 440 Hz. The soundboard is modeled as an orthotropic clamped plate with reinforcing beams and a circular cutout forced by harmonic point loads representing the strings. The model requires the plate outline to be axisymmetric. There are no limitations on the number or orientation of the reinforcing beams. Material property variations, brace cross section, brace pattern layouts, individual brace tapers, plate thickness, and varnish and glue line effects can be studied with this model. A suitable database of material properties and guitar geometry parameters, string lengths, bridge placements, etc. has been built to support the model.

9:35

**BBB4. A model analysis of force-deformation characteristics of a piano hammer.** Hideo Suzuki (Ono Sokki Co., Ltd., 2-4-1 Nishishinjuku Shinjuku-ku, Tokyo 163, Japan)

Effects of a geometry and a Young's modulus profile of a piano hammer on its static force-deformation characteristics are studied using a finite element mode. The material properties of the model are assumed to be linear as well as isotropic. Therefore, only a geometric nonlinearity is considered. A hammer model with a representative geometry and Young's modulus profile of an actual bass hammer has a force ( $f$ )–deformation ( $d$ ) relationship approximated by a function,  $f = Ad^B$ , with ap-

proximately  $B = 1.7$ . On the other hand, the degree of nonlinearity ( $B$ ) of a particular hammer obtained from a dynamic measurement is approximately 2.3 or larger. A hammer with a larger increase of Young's modulus in the inward direction has a larger degree of nonlinearity. The geometry of a hammer has a large effect on the value  $A$  in the formula shown above. A hammer with a shorter height (distance from the top of the hammer to the top of the wood piece) or with a larger width has a larger value of  $A$  (in other words, a larger stiffness).

9:50

**BBB5. An alternate fairy tale for brass instruments.** R. Dean Ayers (Department of Physics-Astronomy, California State University, Long Beach, CA 90840)

Arthur Benade's well-known paperback *Horns, Strings, and Harmony* includes a didactic fairy tale about a bugler forced to play on a uniform pipe and stay in tune with ordinary bugles. With a bit of sawing and some impressive lipping, he manages to succeed. This story fits nicely with the prevailing view that the bell on a brass instrument causes a considerable variation in the effective length of a (singly) closed pipe, thus making the performer's job much easier. Another possible ending for this fairy tale is that the bugler solves his tuning problems with the hand-stopping technique of natural horn players, rather than with his embouchure. One day he accidentally closes the pipe completely and discovers that he now has a perfectly good harmonic series. A slight lengthening of the pipe puts him in tune with his colleagues. This ending is consistent with a view just emerging from time-domain studies that, in terms of reflection back to the driver, the bell of a brass instrument mimics the closed end of a uniform pipe. Demonstrations will be included.

10:05

**BBB6. Influence of bore shape on the resonant frequencies and spectral envelope of a French horn.** J. Duane Dudley and William J. Strong (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

The harmonicity of the resonant frequencies, and the shape of the spectral envelope of the input impedance curve, are two important indicators that might be used in predicting the usefulness of a given air column as a brass wind instrument. These can be determined by approximating the air column as a large number of concatenated lossy cylindrical increments, and computing and plotting the input impedance of the configuration as a function of frequency. In this paper the results of such calculations are described for a number of bore shapes related to the French horn, and the effects of changing various dimensional parameters are illustrated. The contributions of the bell, the leadpipe, and the mouthpiece of a typical open B-flat horn are first examined. Then the horn is idealized and represented by 14 dimensional parameters, so that each can be varied independently to study its effects on harmonicity and spectral envelope, and an optimum configuration is determined. Finally, the effects of inserting the hand into the bell are examined.

10:20

**BBB7. Wind instrument transfer responses.** James W. Beauchamp (School of Music, University of Illinois at Urbana-Champaign, Urbana, IL 61801)

Mouthpiece and output pressure signals from several instruments (e.g., trombone, clarinet, saxophone) have been recorded, harmonic spectra have been computed, and transfer responses have been estimated by taking ratios of corresponding harmonic amplitudes. Each transfer response appears to be a series of points on a smooth high pass filter function, even though it actually corresponds to points on (tending toward the minima of) a multiresonant response curve. According to simplified assumptions the measured responses would be independent of amplitude and fundamental frequency. However, there is, in fact, a considerable degree of variation. Results will be presented and reasons for the variations will be discussed.

10:35

**BBB8. On the bore-to-driver feedback in double-reed conical woodwinds.** J. Agulló, A. Barjau, J. Martínez, and S. Cardona (Department of Mechanical Engineering, Politechnical University of Catalunya, Diagonal, 647, 08028 Barcelona, Spain)

The nonlinear behavior of double-reed drivers makes advisable the use of time domain to study bore-to-driver feedback. The characteristic trends of the impulse response of a conical woodwind are revised. The low intensity of the reflection coming from the first open hole shows the essential role that Bernoulli pressure plays in the reed self-sustained oscillation (also proved by the reed squeaks that players produce to check reed readiness). Time-domain study of these instruments leads to the convolution product relating pressure and velocity at the bore input section. The use of the impulse response  $h(t)$ , the plane-wave and the spherical-wave reflection functions,  $R_p(t)$  and  $R_s(t)$  respectively, as kernel functions in this product is revised. The sharp fluctuations of Bernoulli pressure may impair the use of  $R_p(t)$  while  $R_s(t)$  is unsuitable due to the usual taper discontinuity in the staple-main body junction of these instruments. Computed and experimental initial transients are given for the case of a "tenora" (Catalan tenor shawm).

10:50

**BBB9. Woodwind instrument simulation in real-time.** Michael Park and Douglas H. Keefe (Systematic Musicology Program, School of Music, DN-10, University of Washington, Seattle, WA 98195)

A real-time implementation of the McIntyre-Schumacher-Woodhouse time-domain description of musical oscillators (1983) has been developed using digital signal processing (DSP) hardware. The system is comprised of a microcomputer with installed DSP board including a digital-to-analog converter. The task is to perform a convolution and solve two simultaneous equations with nonlinearities for the unknown pressure and volume flow at successive instants of time. A table lookup with linear interpolation simplifies the original formulation. The reflection functions used in the convolution are calculated based upon analytical models of woodwinds, but the nonlinear valving mechanism is somewhat idealized. One can "play" the instrument via an MIDI keyboard controller. Each MIDI note-on message loads a different reflection function into the DSP board memory, while the pitch wheel controls "blowing pressure." Recorded examples will be presented.

11:05

**BBB10. Patterns of expressive timing in performances of a Beethoven Minuet by nineteen famous pianists and one computer.** Bruno H. Repp (Haskins Laboratories, 270 Crown Street, New Haven, CT 06511-6695)

Onset-to-onset durations of quarter notes and sixteenth notes were measured in nineteen complete performances of the third movement of Beethoven's Piano Sonata op. 31, No. 3. These measurements were compared to those from a computer performance incorporating the "Beethoven pulse" expressive microstructure devised by Manfred Clynes [U. S. Patent No. 4,704,682 (1987)]. This pulse is a particular pattern of relative note durations (and amplitudes) within successive time units, and it is designed to be invariant except for occasional larger deviations at structurally salient points. Although patterns resembling the Beethoven pulse could be found locally in the human performances, on the whole there was little evidence for a fixed, pulselike pattern of timing deviations. Rather, the measurements suggest that great artists change the patterns of relative note durations continuously to meet the local expressive requirements of the composition, and that, despite considerable individual differences, there is some consensus about what these requirements are. If the invariant Beethoven pulse enhances the expressiveness of the computer performance, as it seems to do [B. H. Repp, *J. Acoust. Soc. Am. Suppl. 1* **81**, S92 (1987)], this is apparently not because it captures aspects of human performance. [Work supported by NIH.]

**BBB11. The role of tambura spectra in drone tunings of North Indian ragas.** Edward C. Carterette, Nazir Jairazbhoy, and Kathryn Vaughn (Departments of Psychology and Music, University of California, Los Angeles, CA 90024)

As part of a program on the role of drone interactions with North Indian ragas played on the sitar, the waveforms of the tambura for the most common drone tuning of the four strings [pa, sa1, sa2, sa' (low sa)] were analyzed. Each of the strings was played with and without juari threads that have the effect of separating the string slightly from the

bridge. The resulting sound is referred to by Indian musicians as a buzz. Without juari the pitch heard is very close to the string fundamental and the partials fall linearly in decibels. With juari the pitch is heard at one or even two octaves above the base pitch and the partials have great power as far as the 20th partial. With juari the 4th, 7th, 11th, or 12th, and the 17th partials have more amplitude than the fundamental. Total harmonic distortion is on the order of 1 to 19 dB without juari but ranges from 12 to 44 dB with juari. The upward spread of energy into higher partials imparts richness to tambura tones, and underlies the use of different drone tunings for different ragas. The pitch and timbre changes with tambura are discussed in terms of the psychoacoustics of complex tones.

FRIDAY MORNING, 20 MAY 1988

GRAND BALLROOM A, 9:00 TO 11:00 A.M.

### Session CCC. Psychological Acoustics VII: Multiear Phenomena and Models

Honor O'Malley, Chairman

Teachers College, Columbia University, 525 W. 120th Street, New York, New York 10027

#### Contributed Papers

9:00

**CCC1. Creating a sound field for manipulating interaural level differences separately or independently of the interaural time differences and vice versa.** George F. Kuhn (Vibrasound Research Corporation, 2855 West Oxford Avenue, Englewood, CO 80110)

Using two or more sources whose source strengths and relative phases can be controlled independently, a sound field can be generated in which the interaural pressure level differences (ILDs) and the interaural time differences (ITDs) can be controlled separately. Thus, for a fixed ILD, a range of ITDs, or vice versa, can be presented to the listener. Thus it appears possible to perform some localization experiments that had previously been confined to lateralization methods using earphones. Some calculated results for interaural time differences and interaural level differences, using two sources, will be presented.

9:15

**CCC2. An adaptation model for source displacement in multiple-source environments.** Gregory H. Wakefield (Department of Electrical Engineering and Computer Science, University of Michigan, Ann Arbor, MI 48109-1109)

In general, the acoustic field at the two ears is not sufficient to support source localization for more than one source. Experiments on localization indicate that the co-occurrence of two sources can lead to substantial source localization error that reflects either the ambiguity present in the acoustic field or some form of bias in the binaural processing of that acoustic field. It is shown that displacement of the perceived lateral position of a narrow-band noise away from the lateral position of a narrow-band interferer [G. H. Wakefield and N. F. Viemeister, *J. Acoust. Soc. Am. Suppl.* 1 76, S91 (1984)] does not lie on the surface of possible source positions supported by the acoustic field. Therefore, it does not appear that anomalous localization can be explained entirely with respect to ambiguities in the acoustic field. Instead, a modification of Stern's position variable is proposed to account for source displacement by introducing a frequency-dependent form of spatial adaptation. Much, but not all, of our displacement data are consistent with an additive form of adaptation with a time constant of 25 ms in which the level of adaptation at a given position does not depend on excitation at other positions. The manner in which the model fails to predict displacement for long-duration sources suggests the presence of a weaker adaptation across position.

9:30

**CCC3. Binaural modulation detection.** D. Wesley Grantham and Sid P. Bacon (Bill Wilkerson Hearing & Speech Center and Division of Hearing and Speech Sciences, Vanderbilt University, 1114 19th Avenue South, Nashville, TN 37212)

A sinusoidally amplitude-modulated (SAM) noise (modulation depth: 1.0) was used to mask another SAM noise, the signal, in a two-interval forced-choice experiment. Signal and masker modulation frequency were both either 4 or 32 Hz, and were added in quadrature. The same wideband noise carrier was used for both masker and signal (spectrum level: 15 dB SPL). The whole stimulus was filtered after modulation (either 4-kHz low pass or 4-kHz high pass). Modulation depth ( $m$ ) of the signal was varied adaptively to track threshold. In the case of diotic presentation (Mo-So), threshold  $m$  was about 0.30 for both modulation frequencies and both filtering conditions. In a binaural condition Mo-Sm (masker presented to both ears and signal to only one ear) some subjects showed a release from masking of up to 10 dB ( $m = 0.10$ ), while others showed none. The release from masking occurred for both filtering conditions, but only for the lower (4-Hz) modulation rate. The results will be discussed in relation to current models of binaural processing and the sluggish nature of the binaural system. [Work supported by NIH.]

9:45

**CCC4. Detection of interaural phase shifts and level differences in portions of broadband noise.** William A. Yost (Parly Hearing Institute, Loyola University, 6525 N. Sheridan Road, Chicago, IL 60626)

Versions of Cramer-Huggins pitch stimuli were generated such that a narrow band (bandwidths ranged from 1-100 Hz) of a wideband, diotic noise (low-passed filtered at 8000 Hz) was presented dichotically with an interaural phase shift or an interaural level difference. The threshold value for detecting a phase shift or a level difference in the dichotic, narrow-band portion of the broadband noise was determined as a function of the center frequency of the dichotic, narrow band of noise and as a function of the bandwidth of the dichotic, narrow band of noise. A single-interval procedure without feedback was used to estimate three point psychometric functions relating  $d'$  to interaural phase or  $d'$  to interaural level difference. A random variation in the interaural level difference of the entire noise waveform was also used to force the listeners to use pitch as the cue for detection rather than some aspect of lateral position. The data will be

10:00

**CCC5. Lateralization on the basis of interaural envelope delays: The effect of component starting phase.** R. H. Dye, A. J. Niemiec, and M. A. Stellmack (Parmly Hearing Institute, Loyola University of Chicago, 6525 N. Sheridan Road, Chicago, IL 60626)

Threshold interaural envelope delays were measured with a 2 AFC paradigm as a function of modulation frequency ( $f_m = 25, 50, 100, 200, 300, 400,$  and  $500$  Hz) for three- and five-component complexes whose center frequencies ( $f_c$ ) were equal to  $4000$  Hz. Comparisons were made between thresholds obtained when the starting phases were randomized and when they were all set to zero. The level of each component was approximately  $50$  dB SPL, and the signal duration was  $200$  ms with  $10$ -ms linear rise/decay times. While reductions in depth of modulation elevate threshold envelope delays [G. B. Henning, *J. Acoust. Soc. Am.* **55**, 84–90 (1974)], no such effects were found for phase randomization even when modulation rates were low and all components fell within a critical band. This seems surprising in light of the fact that both reduced depth of modulation and phase randomization reduce the peak factors of the waveforms, which is commonly believed to interfere with entrainment by the peripheral auditory system. Efforts to reconcile the absence of phase effects with the deleterious effects of reduced depth of modulation will be presented. [Work supported by NIH/NINCDS and AFOSR.]

10:15

**CCC6. Accruing binaural cues across frequency.** Thomas N. Buell and Ervin R. Hafter (Psychology Department, University of California, Berkeley, CA 94720)

Localizing a complex sound requires in part an analysis of the interaural difference of time (IDT) of the component frequencies. Above about  $500$  Hz, the IDT is constant across the frequency spectrum. This commonality may be used to group components that arise from a single source as well as segregating such components from those which arise from competing sources. Interactions across components can be shown by measuring changes in IDT thresholds when a target is lateralized in the presence of other simultaneous sounds. When the nontarget sound has a dissimilar IDT, thresholds may actually increase. This effect has been aptly termed "interference" [D. McFadden and E. G. Pasanen, *J. Acoust. Soc. Am.* **59**, 634–639 (1976)]. In the present experiment, listeners detected an IDT in a  $600$ -Hz target tone both with and without the presence of a nontarget tone. Of special interest was the harmonic relation between target and nontarget. Among other factors, the results show that a simple relation such as  $400$  Hz can produce more interference than one that is nonharmonic, even when the interfering tone is closer to the target, e.g.,  $531$  Hz. [Research supported by NIH.]

**CCC7. Discrimination of direction in the free field.** Ervin R. Hafter, Thomas N. Buell, David A. Basiji, and Elizabeth E. Shriberg (Department of Psychology, University of California, Berkeley, CA 94720)

The psychophysics of direct comparison (two-alternative, forced choice) was applied to the study of localization. For complex signals, the interaural parameters presented by stimuli in the free field are more "natural" than those heard through earphones though more difficult to control. In this study, digital preprocessing was used to reverse the spectral characteristics of each loudspeaker to make its response uniform and indiscriminable. The listener sat in an anechoic chamber with his or her head optically fixed at the center of a semicircular array of speakers. The stimuli were trains of  $n$  filtered clicks ( $n = 1, 9,$  or  $18$ ) whose center frequencies were either  $2050, 4050,$  or  $6050$  Hz. Inter-click intervals (ICIs) were either  $4$  or  $12$  ms and levels were set so that a single click was  $10$  dB SL. Minimum audible angles (MAAs) were found to increase as the azimuth increased from  $0$  to  $60$  deg; performance was best at  $2050$  Hz. Studies of lateralization with earphones have led to speculations about "binaural adaptation," a process whereby improvements in threshold with increased  $n$  depend on the click rate ( $1/ICI$ ). Similar analyses show this also to be true for free-field listening. [Research supported by the NIH.]

10:45

**CCC8. Sound localization without interaural time or intensity differences.** Frederic Wightman, Doris Kistler, Marianne Arruda, and Linda Schwister (Department of Psychology and Waisman Center, University of Wisconsin, Madison, WI 53706)

In a series of localization experiments, blindfolded listeners were asked to give the apparent azimuth and elevation coordinates of  $250$ -ms broadband noise bursts. The noise bursts were spectrally shaped so that the spectrum level in each critical band was set randomly (from trial to trial) within a  $20$ -dB range. In a reference condition, the sounds were transduced by miniature loudspeakers placed at  $36$  locations in an anechoic chamber. Performance in this free-field condition agreed with previous results in the literature except for an increased frequency of front-back reversals, which is felt to be the result of the lack of a visual frame of reference. In a control condition, digital techniques were used to synthesize headphone-presented stimuli that were nearly identical, as measured in listeners' ear canals, to the free-field stimuli. Localization performance with these stimuli was virtually the same as in free field, thus verifying the adequacy of the simulation. The digital filters used to synthesize the stimuli were then modified such that in one condition the interaural time differences were removed from all  $36$  stimuli, and in another condition the interaural amplitude differences were removed. In both of these conditions, localization performance was degraded. However, preliminary results suggest that the degradation caused by removal of interaural time differences is considerably greater than that caused by removal of interaural intensity differences. [Work supported by NIH, NSF, and NASA.]