

## Prediction of noise in ships by the application of “statistical energy analysis.”

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## PROGRAM OF

### The 98th Meeting of the Acoustical Society of America

Hotel Utah • Salt Lake City, Utah • 26–30 November 1979

TUESDAY MORNING, 27 NOVEMBER 1979 BONNEVILLE ROOM 4, 9:00 A.M. TO 12:05 P.M.

#### Session A. Engineering Acoustics I: General

Mauro Pierucci, Chairman

*Department of Aerospace and Engineering Mechanics, San Diego State University,  
San Diego, California 92182*

Chairman's Introduction—9:00

#### Contributed Papers

9:05

**A1. The Green's function of the fluid-loaded plate.** D. G. Crighton (Department of Applied Mathematical Studies, University of Leeds, Leeds, England)

The surface response of an infinite thin elastic plate or membrane to concentrated forcing is considered under the (realistic) condition that the normalized response is specified by three parameters: (1) a ratio  $\Omega$  of driving frequency to the vacuum coincidence frequency, (2) the product  $X$  of distance from the excitation point and the vacuum free wavenumber, and (3) a parameter  $\epsilon$  characterizing fluid loading at coincidence. We use the fact that in all applications  $\epsilon$  is small to construct matched asymptotic expansions for the Green's function covering the entire range of frequency and distance ( $0 < X, \Omega < \infty$ ). The method is applicable equally to line or point forcing and to plate or membrane response, significant differences being confined to the nearfield. The farfield for  $\Omega = O(1)$  consists of free surface or leaky waves plus an acoustic field decaying as  $X^{-1/2}$  at moderate  $X$  and as  $X^{-1}$  at very large  $X$ , with a transition expressed by parabolic cylinder functions; the description for other  $\Omega$  is obtained, but is more complicated.

9:20

**A2. The saga of the plate–fluid interaction. II.** Mauro Pierucci (Department of Aerospace and Engineering Mechanics, San Diego State University, San Diego, CA 92182)

Over the last two years, the author [J. Acoust. Soc. Am. **65**, 1190–1197 (1979)] and other researchers have reanalyzed the plate–fluid interaction problem from different perspectives. In this paper the eigensolutions to the fluid-loaded thick plate equation are presented as a function of modulus of elasticity fluid loading and structural damping. The thick plate equation allows for the inclusion of shear deformation and rotary inertia effects on the vibrational response of the plate. Solutions for loci of the different modes as a function of frequency are presented in the fluid wavenumber plane. It is shown that under certain conditions the roots of the system cross from one Riemann sheet to the other thus changing the basic characteristics of the plate response and also of the radiated acoustic field. Internal structural damping is also shown to have the same effect upon the behavior of the root loci. Some of the phenomena discussed are a result of the rotary inertia terms in the plate equations

and, therefore, cannot be explained by the use of the simpler thin plate analysis.

9:35

**A3. Random vibration field on a plate.** R. V. Waterhouse (Department of Physics, The American University, Washington, DC 20016)

The properties of a steady-state random, or reverberant, sound field in a reverberation chamber with stationary boundaries are fairly well understood. When the driving signal consists of a single-frequency tone, and many modes are excited, the sound field in the chamber becomes a randomly irregular interference pattern whose statistical properties follow known distributions. For example, the mean-square pressure values, suitably sampled, follow an exponential distribution, the simplest kind of gamma distribution. It is pointed out that the same gamma distributions held for the vibrational field on a thin elastic plate vibrating in flexure under the same conditions of pure tone and multimode excitation. Knowing the distribution of the acceleration values on such a plate is of practical value as it allows one to estimate the confidence levels of the average of a limited number of sample values.

9:50

**A4. *In situ* determination of loss and coupling loss factors by the power injection method.** David Alan Bies and Shahul Hamid (Department of Mechanical Engineering, University of Adelaide, Adelaide, South Australia, 5000)

Inversion of the linear power balance equations for a two coupled plate model is used to determine the plate loss factors and the coupling loss factors *in situ*. To accomplish the determinations, injected power at points chosen at random was measured. To ensure effective statistical independence of modes, each plate was driven at five different points in sequence and the response of both plates in each case was determined at ten randomly chosen points. Good agreement is obtained between the predicted and measured coupling loss factors, and between the *in situ* loss factors and loss factors determined for each plate separately also in steady state from power injection measurements. Loss factors determined by transient decay methods are consistently lower than those determined by either steady-state method. During reverberant decay, the more lightly

damped modes predominate, giving rise to an apparent loss factor which is significantly less than the steady-state loss factor determined under conditions of essentially equal energy per mode.

10:05

**A5. Use of the Prony method in reflection coefficient measurements of acoustic panels.** D. H. Trivett (Underwater Sound Reference Detachment, Naval Research Laboratory, P.O. Box 8337, Orlando, FL 32856) and A. Z. Robinson (Oak Harbor Marine Associates, Southport, FL 32409)

A modified Prony method is presented for measuring steady-state reflection coefficients of acoustic panels. The method extrapolates the steady-state amplitude from the transient portion of the signal allowing time limited measurements. This method is applied to measurements of square panels 76 cm on an edge and 0.95 cm thick of steel, aluminum, and Lucite in the frequency range of 3–10 kHz. The signals were time limited to  $200\ \mu\text{s}$  ( $0.6\text{--}2.0\ \lambda$ ) by the arrival of the diffracted signal from the panel edges. Results are compared with theoretical values and indicate that the method is capable of making measurements subject to ambient noise. [Work supported by Target Strength Reduction Program managed by DTNSRDC of NAVSEA 037.]

10:20

**A6. On the compressional damping mechanism in the free impedance response of damped three-layer beams.** B. E. Douglas (David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402)

The transverse free mechanical impedance response of an elastic-viscoelastic-elastic beam incorporating the compressional damping mechanism is considered. The work of Douglas and Yang is extended to include shear deformation and rotary inertia in the elastic layers. The effects of nearly incompressional viscoelastic damping cores on the compressional frequency and hence on the spectral range of damping effectiveness for the composite is also discussed. Results of the analysis are shown to compare favorably with experimental results for a damped three-layer beam which was optimized for compressional damping and in which the influence of the shear damping mechanism was intentionally minimized.

10:35

**A7. Determination of sound pressure distributions in liquid-filled, elastic and viscoelastic tubes by finite-element method.** E. A. Schroeder and L. T. Ho (David W. Taylor Naval Ship Research and Development Center, Bethesda, MD 20084)

Based on the finite-element approach, the elastic tube was represented by the conical shell elements CONEAX, while the fluid was modeled by the solid elements TRAPAX. Matrix equations at grid points for the displacement vectors in the solid and sound pressure field in the fluid were coupled at the fluid-structure interface. Constraints were applied and material parameters were chosen so that the axial displacements at grid points in the fluid would correspond to sound pressures that satisfy the wave equations. The coupled equations were solved for sound pressures, using structure computer program NASTRAN. Excellent agreement was obtained between the calculated and experimental results of low-frequency sound pressure distributions in an elastic tube. For the case of a liquid-filled, rigid tube lined internally with viscoelastic material, both the fluid and the liner elements were represented by the trapezoidal ring elements TRAPRG. Good correlation was observed qualitatively between the calculated and experimental sound attenuation in a viscoelastic tube. [Work supported by Naval Sea Systems Command.]

10:50

**A8. Velocimeter for quality control monitoring in rubber production.** R. D. Corsaro, J. Jarzynski, L. Burns, and W. B. Moniz (Naval Research Laboratory, Washington, DC 20375)

A velocimeter was designed for longitudinal sound speed measurements on 3-mm-thick (ASTM) rubber samples. Batch-to-batch variations in sound speed are detected to 0.1% precision by operators without technical experience. Each measurement required five minutes with the prototype device, and substantial reductions are expected with minor design modifications. The device consists of two ultrasonic transducers with a rubber sample sandwiched between them. During device assembly, the sample is squeezed by the transducers such that a known and reproducible transducer-transducer separation distance is obtained. A 5-MHz cw signal is then applied to one transducer. The phase angle difference of the signal received by the other transducer is read on a meter face, and is related to the wave speed in the rubber. By performing a detailed theoretical analysis of this geometry, we are able to meaningfully compensate the value of the meter reading for variations in temperature, thickness (related to the degree of sample squeezing used), and sound absorption, and also use the device for absolute sound speed and attenuation measurements.

11:05

**A9. Stoneley waves at an imperfectly bonded interface.** Michael Schoenberg (Schlumberger-Doll Research, P.O. Box 307, Ridgefield, CT 06877)

A model for an imperfectly bonded interface between two elastic media has been proposed. Displacement across this surface is not required to be continuous but it is taken to be linearly related to the stress traction which is continuous across the interface. For isotropic interface behavior, there are two independent interface compliances,  $\eta_N$  and  $\eta_T$ , where the component of the displacement discontinuity normal to the interface is given by  $\eta_N$  times the normal stress and the parallel component is given by  $\eta_T$  times the shear stress. Stoneley wave velocities (possibly complex) at such interfaces between two elastic halfspaces are calculated and are found to be dispersive even when the compliances,  $\eta_N$  and  $\eta_T$ , are real and frequency independent. When the interface is purely dissipative such that the stress traction is proportional to the velocity discontinuity, the compliances are pure imaginary and proportional to the inverse of frequency yielding complex, but in this case, nondispersive, Stoneley wave velocities.

11:20

**A10. Experimental studies of ultrasonic radiation from cylindrical shells in water.** K. Fritsch, C. W. Allen, and E. F. Carome (John Carroll University, Cleveland, OH 44118)

Studies are being made of ultrasonic waves radiated into water from air-filled cylindrical metal shells. A shell is directly excited by polyvinylidene fluoride thin film transducers mounted on its inside surface. By using arrays of such transducers it has been possible to excite various normal modes of the cylinder. Both schlieren and acoustic probe techniques are being used to examine the radiated acoustic fields. Data are presented to indicate the applicability of this new technique. [Work supported in part by ONR.]

11:35

**A11. Frequency-wavenumber spectral analysis of borehole acoustic waves.** T. L. Marzetta, D. Leslie, M. Schoenberg, and J. Aron (Schlumberger-Doll Research, P.O. Box 307, Ridgefield, CT 06877)

In a laboratory experiment an acoustic pulse excited waves in a fluid-filled borehole that had been drilled in a solid block of epoxy.

Typical wavelengths were the same order of magnitude as the hole radius. Waveforms were received at many different ranges and conventional and high resolution, maximum likelihood methods (MLM) of frequency-wavenumber ( $\omega, k$ ) spectral analysis are applied to this data. The spectra obtained are compared to theoretical dispersion calculations. Good agreement is obtained. Encouraging improvements in wavenumber resolution over conventional spectral estimation are obtained by the use of the MLM techniques.

11:50

**A12. Surface intensity patterns for a baffled plate.** J. Daniel Brito (Deere & Company Technical Center, Moline, IL 61265)

The spatial characteristics of the acoustic power flow at the surface of a baffled simply supported rectangular plate vibrating in flexural motion are studied to obtain an insight on the process of energy exchange between the plate and the acoustic medium. An expression for the local radiation efficiency as a function of space and frequency is presented and used to compute sound intensity patterns at the surface of the vibration plate, under different types of excitation of the plate-bending motion. The space average of the local radiation efficiency is compared with published values and with measurements by the reverberant room method. Frequency averages ( $\frac{1}{3}$  octave band) of this local radiation efficiency at points on the plate are compared with measurements by the accelerometer-microphone method. Satisfactory agreements are obtained in all these comparisons. [Work performed at M.I.T.]

**TUESDAY MORNING, 27 NOVEMBER 1979 BONNEVILLE ROOM 5, 9:00 A.M. TO 12:11 P.M.**

## **Session B. Noise I and Underwater Acoustics II: Shipboard Noise Control**

**Louis A. Herstein, III, Chairman**

*Ship Silencing Division Code 037, Naval Sea Systems Command, Washington, D.C. 20362*

**Chairman's Introduction—9:00**

### ***Invited Papers***

**9:05**

**B1. A personal philosophy of design for noise control in ships.** J. R. Baylis (5820 Westchester Street, Alexandria, VA 22310)

This personal philosophy has evolved from eight years of active participation in the research and the design of Navy ships for controlled noise characteristics. The design for the control of noise begins with the establishment of requirements. Noise specifications must serve an understandable operational purpose that can be communicated to the many people that will design and build the ship. The requirements must be capable of being met within the state of the art, and at a cost that is commensurate with the value of the noise control. The cost must be acceptable to the producer as well as the customer, and the producer must understand from the beginning the amount of engineering and test that he is committed to. The techniques for controlling the noise of ships are well known, but there are choices to be made for those techniques to be used in construction, and those that can be used to come up to specification after test. An allowance should be made for the cost of test and for the cost of corrections to be made after test.

**9:30**

**B2. U.S. merchant ship noise: Recent measurements and proposed criteria.** R. S. Gales (Naval Ocean Systems Center, San Diego, CA 92152) and D. T. Jones (United States Coast Guard, Washington, DC 20590)

To provide guidance in the regulation of noise on U.S. merchant ships, noise levels and octave-band spectra were measured on seven ships of the U.S. Merchant Fleet. Ships selected included tankers, ore carriers, general cargo, and container ships; with steam turbine, gas turbine, and diesel propulsion. They ranged in size from 8000 to 36 000 gross tons and in construction date from the 1920's and 1970's. Locations measured included machinery and engine rooms, passageways, berthing spaces, mess and recreation spaces, bridge, radio room, and offices. Noise exposures based on measured sound levels and time spent in various spaces were calculated for individual crew members in various occupational categories. Exposures are compared with various criteria for hearing conservation, including the 24-h equivalent continuous sound level ( $L_{eq} 24$ ). A-weighted sound levels in other ship spaces are compared with various function-based criteria, such as speech communication, hearing of warning signals, rest and relaxation, and recovery from temporary hearing loss. Recommended criteria for A-weighted sound level on U.S. Merchant vessels are presented for immediate use and future goals.



**B3. Prediction of noise in ships by the application of "statistical energy analysis."** John Ødegaard Jensen (The Acoustics Laboratory, Technical University of Denmark, DK-2800 Lyngby, Denmark)

If it will be possible effectively to reduce the noise level in the accommodation on board ships, by introducing appropriate noise abatement measures already at an early design stage, it is quite essential that sufficiently accurate prediction methods are available for the naval architects. In general, the structure-borne noise contribution from the various noise sources may be precalculated with a reasonable accuracy using empirically based calculation models. The prediction very often fails, however, when the empirically based calculation model is applied for an untypical structure or for a special noise abatement measure, e.g., increased structural damping. The paper discusses whether it might be possible to derive an alternative calculation model based on the "statistical energy analysis" approach (SEA). By considering the hull of a ship to be constructed from plate elements connected by combination of  $L$  junctions,  $T$  junctions, and cross junctions, a SEA-calculation model has been derived. Examples on application of the SEA model for prediction of the structure-borne sound transmission are given, partly through simple two-element structures consisting of stiffened and unstiffened plate panels, partly through a hull section consisting of several stiffened plate sections. The results of the SEA calculations are compared with corresponding results of vibration measurements on the structures.

10:20

**B4. The airborne path in shipboard noise control.** M. C. Junger (Cambridge Acoustical Associates, Inc., 54 Rindge Avenue Extension, Cambridge, MA 02140)

Reverberant noise in living and working spaces other than the machinery room is typically controlled by structure-borne paths. However, as machinery vibration isolation and foundation damping techniques are perfected, airborne flanking paths discussed in this paper become increasingly important. In contrast to representative masonry walls, the finite extent of shipboard partitioning and bulkheads is an essential factor in determining transmission loss (TL). Consequently, structural damping is beneficial not only near coincidence, but also at higher frequencies. A second distinguishing feature of much shipboard partitioning is frame stiffening. This extends the beneficial effect of damping to frequencies below coincidence. Stiffening frames cause marked TL degradation below the TL of the underlying uniform plate, in spite of the latter's smaller mass. In conclusion, the concept that mass per unit area is the single most important factor in determining TL is not relevant to the shipboard situation.

10:45

**B5. Current practices in shipboard airborne noise control and future needs.** Daniel L. Nelson (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138)

In recent years, shipboard airborne noise criteria have been applied much more stringently than in the past. Further, noise criteria are likely to be lowered for future ship designs. In response to this, more noise sources and transmission paths need to be considered in the acoustical design process. This paper considers noise control treatments which are presently coming into accepted practice in this country and discusses the merits as well as liabilities of these treatments. Needs for future research and development efforts are also presented in light of more restrictive noise criteria.

11:10

**B6. Airborne noise reduction of surface ship propulsion gearboxes and forced draft blowers.** Edward V. Thomas (Machinery Isolation Branch, Code 2742.2, David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402)

Five surface ships were treated with a propulsion gearbox two-phase cladding-damping treatment. The effects of treatment installation procedures on airborne noise reduction were evaluated as a function of ships speed. Two surface ships were treated with developmental antireflective airborne noise treatments to reduce forced draft blower noise levels. A resonant sound absorber was installed in another ship to reduce forced draft blower noise.

### *Contributed Papers*

11:35

**B7. Control of shipboard airborne noise caused by propeller cavitation with a naturally aspirated compliant air layer.** Leslie M. Gray (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138)

Propeller cavitation can often be a severe source of shipboard airborne noise, especially in after spaces on high-speed or high-

powered vessels. Control of this noise source is difficult since the cavitation imparts vibration energy to major hull structure. Typical noise control of this source usually involves such treatments as damping, isolated decks and bulkheads, and added mass. Unfortunately, this additional weight is often unacceptable to the ship designer. This paper investigates control of this source by introduction of a compliant air layer between the water and the hull

structure, which will act as a simple springlike isolator. The potential effectiveness of this compliant air layer is outlined. A source of air is proposed which employs the negative pressure reduction due to flow over an emitter on the hull to suck atmospheric air into the ship's boundary layer. An example design is presented for a high-speed yacht which will provide adequate air for substantial reduction of propeller-airborne noise for vessel speeds over 15 knots.

11:47

**B8. Structure-borne noise transmission through ship stanchions.** C. B. Burroughs and R. W. Fishcer (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138) and F. L. Shiflet (Hopeman Brothers Inc., Waynesboro, VA 22980)

Stanchions provide a path for structure-borne noise transmission from a noise source space on one level to noise sensitive spaces on another level in commercial and combatant surface ships. Measurements of structure-borne noise from one level to an adjacent level were made with and without a stanchion in place in the laboratory full-scale model of ship compartments at Hopeman Brothers. Measured data of the structure-borne noise transmission loss

across the stanchion and the difference in the airborne sound power levels radiated into the compartments with and without the stanchion in place are presented. Results of statistical energy analysis (SEA) are compared to measured data. Methods for reducing structure-borne noise transmission through stanchions are suggested.

11:59

**B9. Airborne noise criteria on naval ships: Estimating crew response.** David R. Lambert (Code 5121, Naval Ocean Systems Center, San Diego, CA 92152)

Graphs are presented for predicting the response of shipboard personnel to airborne noise when a given compartment type just meets a proposed A-weighted criterion. The graphs were constructed from the responses of personnel in selected compartments aboard eight U.S. Naval ships. The author used the responses to questions on habitability to rate each compartment on a five point scale ranging from satisfactory to unsatisfactory. These ratings were then plotted as a function of A-weighted sound pressure level, and parameters for the curves estimated. [Work sponsored by the Naval Sea Systems Command.]

**TUESDAY MORNING, 27 NOVEMBER 1979 BONNEVILLE ROOM 3, 9:00 A.M. TO 12:20 P.M.**

## Session C. Physical Acoustics I: Acoustic Remote Sensing I: Atmospheric Science

Edmund H. Brown, Chairman

*NOAA Wave Propagation Laboratory, Boulder, Colorado 80302*

Chairman's Introduction—9:00

### Contributed Papers

9:05

**C1. The research and application of acoustic radar systems in Australia.** N. A. Shaw and I. A. Bourne (Footscray Institute of Technology, and University of Melbourne, Melbourne 3011, Australia)

Research and development of atmospheric acoustic radar systems have been closely supported by "application" programs associated with industrial needs. To study this interaction, a survey was conducted on (a) the nature and emphasis of acoustic research programs, (b) the degree of successful industrial application of various systems, and (c) the current and anticipated needs of industry for improved acoustic radar performance. Institutions involved in researching atmospheric acoustic radars include universities, colleges of advanced education, and defense establishments. The power, mining, and larger manufacturing industries have used acoustic probes mainly as a tool for either evaluating meteorological parameters conducive to air pollution problems or as part of extensive environmental impact studies. The somewhat qualitative records by single axis systems are still susceptible to interpretation difficulties. Classification of data, at least into categories and formats suitable for further digital analysis, has been sought, but now the greatest interest exists in multi-axis systems which are thought to be capable of measuring wind profiles and dispersion parameters to heights of 1000 m as well as retaining the conventional facsimile display.

9:20

**C2. Sodar studied atmospheric temperature inversions of the Salt Lake Valley.** G. Davis and D. R. Dickson (Department of Meteorology, University of Utah, Salt Lake City, UT 84112)

Inversional temperature phenomena indigenous to Utah's Salt Lake Valley were observed and studied over a continuous time span of 28 days occurring in December–January 1976–1977. The phenomena were combined with local topographical barriers to maintain a situation of marked atmospheric stability in the lowest kilometer. The nonturbulent conditions resulting were commonly neutral in character, but composite conditions retained generally stable patterns of inversion genesis and disintegration. Particular attention was given to comparing available rawinsonde raw data (soundings) with their coincidental sodar counterparts obtained from the continuous record. Results implied a compatibility between two techniques of sensing used, one being a remote sensor, the other *in situ*. Best agreement was discovered in the inversion cases below 500 m and with those of stronger intensities.

9:35

**C3. Estimates of surface heat flux from sodar and laser scintillation measurements in the unstable boundary layer.** R. L. Coulter and M. L. Wesely (Atmospheric Physics Section, Radiological and Environmental Research Division, Argonne, IL 60439)

Measurements of acoustic backscatter in the lower planetary boundary layer and optical line-of-sight scintillation in the surface layer are used separately to compute the sensible heat flux in the unstable surface layer. Comparisons with simultaneous low-level point measurements by eddy correlation show good agreement both in absolute magnitude and response to rapid changes, indicating that acoustic methods may be successful over less homogeneous terrain where conventional surface layer measurement techniques

are often inaccurate. Corrections to take into account the effects of humidity fluctuations are found necessary in order to achieve accuracies within 10%. Free convection is assumed to permit interpretation of the sodar data, while either forced or free convection is assumed for the scintillation data. A systematic overestimate of surface heat fluxes is found from acoustic measurements made during the morning when the convectively mixed layer is rapidly increasing in height. [Work supported by U.S. Dept. of Energy and U.S. Environmental Protection Agency.]

9:50

**C4. Preliminary results on observations of chinooks using a Doppler acoustic sounder.** S. A. Leelananda, R. B. Hicks, P. J. Irwin and T. Mathews (Department of Physics, The University of Calgary, Calgary, Alberta, Canada T2N 1N4)

An acoustic sounder array located on a rural site 20 km west of Calgary is described. The array is designed to sense atmospheric turbulence with a working vertical range of 2 km and also to provide information on the vertical wind profile up to 800 m above the site. The array consists of a central, vertically directed, monostatic array (output acoustic power ~ 100 W), and four passive inclined receivers, each acting as a bistatic receiver for the central monostatic transmitter. The bistatic receivers are arranged along two mutually perpendicular horizontal baselines. On-line microprocessors are used to retrieve Doppler shift data from all receiver channels, and hence to calculate the wind profile. Typical results from the Doppler acoustic sounder array are presented. It is hoped that by the time of presentation, soundings will have been made during at least one Chinook event (a foehn-type wind associated with airflow over the nearby Rocky Mountains).

10:05

**C5. Comparison of acoustic scattering data with predictions from a second-order numerical model of the statistically stable boundary layer.** W. D. Neff (NOAA, Wave Propagation Laboratory, Boulder, CO 80302)

We have utilized a simplified second-order turbulence closure scheme to predict acoustic scattering profiles as a function of time from a numerical model of the atmospheric boundary layer under statically stable conditions. Comparisons have been made with data obtained at South Pole Station, Antarctica. Two cases in particular were analyzed. The first was that of warm air advection over a cold surface. The second involved the thin boundary layer and elevated scattering layers that form when the geostrophic pressure gradient force opposes the thermally driven slope winds characteristic of the East Antarctic ice sheet.

10:20

**C6. Atmospheric finestructure in an arctic valley.** B. R. Kerman and R. E. Mickle (Boundary Layer Research Division, Atmospheric Environment Service, 4905 Dufferin Street, North York, Ontario, Canada, M3H 5T4)

Multiple layers of turbulence were observed by an acoustic sounder in the Shaskwak Valley of the southern Yukon. The layers, extending in time for more than five days, were found to correlate highly with temperature steps observed by a tethered balloon system. During the period of the observation, the wind was virtually calm and the skies clear, suggesting a radiational cause for the phenomenon. Various mechanisms are discussed with particular attention to the possibility of a dynamic instability of katabatic (Stewartson) sidewall layers. Order of magnitude estimates suggest that the lateral turbulent heat flux of such eddies is comparable to the radiational heat losses of the valley walls.

10:35

**C7. An intercomparison of acoustic Doppler systems.** J. C. Kaimal (NOAA, Wave Propagation Laboratory, Boulder, CO 80302)

During late August and early September 1979 the Boulder Atmospheric Observatory (BAO) was the site of an international comparison of low-level sounding techniques. Called the Boulder Low-Level Intercomparison Experiment (BLIE), its objective was to compare a wide range of sounding systems including free-rising sondes, tethered sondes, tower-mounted sensors, remotely piloted aircraft, and acoustic and electromagnetic remote sensors. Six acoustic Doppler systems were compared. This paper will describe procedures used for the intercomparison and offer some preliminary results showing how the systems compared with one another and with *in situ* wind measurement on the 300-m BAO tower.

10:50

**C8. Monitoring the diffusivity of the atmospheric boundary layer by acoustic sounding.** B. R. Kerman (Boundary Layer Research Division, Atmospheric Environment Service, 4905 Dufferin Street, North York, Ontario, Canada, M3H 5T4)

The concept of asymptotic diffusion states is extended to the outer boundary layer. Various models are introduced for the determination of maximum ground level concentration based on classical similarity arguments for plume rise, equilibration, and downwash. An analysis is offered for the estimation of the controlling mean and turbulent parameters (surface heat flux and stress, mixing depth, wind speed and outer flow stability) by acoustic sounding. Further developments are outlined for the estimation of internal turbulent parameters based on revision of the classical Monin-Obukhov similarity theory in terms of acoustic scattering observables.

11:05

**C9. Analysis and prediction of the planetary boundary layer using an integrated sodar-model measurement system.** D. W. Thomson and M. Logan (Department of Meteorology, Pennsylvania State University, University Park, PA 16802)

The surface concentration at a given location of passive pollutants emitted from surface or elevated sources depends upon the effective source height and the environmental vertical wind and temperature profiles. By automated sequential application of widely used boundary-layer inversion rise and structure, and plume rise and diffusion models in the signal processing computer of the Penn State Doppler Sodar System, measurement and modeling functions have been combined. Thus, the computer outputs not only current meteorological conditions but also predicted boundary-layer development and plume characteristics. Since environmental impact data rather than raw meteorological measurements are output, application of the system for techniques such as supplementary control system technology is greatly facilitated. The hardware and software design of this unique acoustic sounding system are presented and a demonstration including a movie are given of its application to a power plant high stack diffusion problem.

11:20

**C10. Studies of the planetary boundary layer using the CRPE triple Doppler sodar.** C. Klapisz, A. Weill, C. Jaupart, O. Taconet, P. Bouzeloup, and L. Eymard (CRPE (CNET/CNRS), 92131 Issy-les-Moulineaux, France)

The triple Doppler sodar is now an operational instrument. In convective situations, it allows us to measure the local vertical heat flux, the vertical velocity variance, and the rate of dissipation of the turbulent kinetic energy, as well as local stresses. It also allows the determination of the local inversion height. We briefly recall

some technical aspects concerning the data treatment. We then discuss the relative importances of the different terms which appear in the turbulent kinetic energy budget and justify the parameterization of several variables introduced. Finally, we review the studies we have undertaken in the homogeneous boundary layer, and present the perspectives offered for the study of the heterogeneous and perturbed boundary layer.

11:35

**C11. The potential importance of correlated humidity and temperature variations for atmospheric acoustic back-scatter.** J. F. R. McIlveen (Department of Meteorology, Pennsylvania State University, University Park, PA 16802)

The dependence of the velocity of sound on temperature and humidity mixing ratio is derived and the commonly applied expression of Wesley (1976) corrected. Variations in the acoustic refractive index produced by turbulent fluctuations in temperature, humidity mixing ratio and vapor pressure or density are examined. The resulting expressions are used in a simple approach to the structure function constant for acoustic refractive index. Wesley's (1976) conclusions (using expressions for structure constants in the surface boundary layer) about the relative importance of terms dependent on humidity structure are confirmed. The importance of the correlation between variations in temperature and humidity for the strength of echoes detected by monostatic sodar is discussed.

11:50

**C12. A comparison of acoustic sounder returns and  $C_T$  estimated from meteorological parameters.** D. A. Smith, R. B. Hicks, and T. Mathews (Department of Physics, The University of Calgary, Calgary, Alberta, Canada T2N 1N4)

A vertically oriented monostatic acoustic sounder has been used in conjunction with a tethered balloon profiler for a series of observations on the stable atmospheric boundary layer immediately following sunrise. Estimates of relative average values of  $C_T$  over the height range 60–220 m were derived from the acoustic backscatter results and were compared with predictions based on wind and temperature profiles using several different models. Our results show that reasonable correlation can be achieved using the temperature gradient as the sole meteorological parameter. Use of more sophisticated models based on similarity functions or atmospheric energy budget does not significantly improve the correlation obtained.

12:05

**C13. The detection of the temperature structure coefficient of atmospheric boundary layer by acoustic radar.** Zhou Ming-yu, Lu Nai-ping, and Cheu Yan-juan (Institute of Atmospheric Physics, Academia Sinica, Peking, China)

The magnitude of temperature structure coefficient is computed by sounder echo data. The results showed that the magnitude of  $C_T^2$  has an obvious fluctuation with time. It has a character of positive bias distribution, and fits the logarithmic normal distribution. The analysis of the spectra  $C_T^2$  showed that, the spectrum  $F(k)$  of  $C_T^2$  usually fits the  $k^{-4}$  rule in stable and unstable stratifications. There exists a peak range of energy on the frequency range  $10^{-1} - 3 \times 10^{-4} \text{ s}^{-1}$ . The analysis of  $C_T^2$  on time-height counter section indicates that the distribution of  $C_T^2$  has horizontal stratum structure in stable stratification and it fits the rule  $C_T^2 \propto z^{-4/3}$  in unstable stratification. These results measured by acoustic radar agree with earlier direct observations by airplanes [L. R. Tsvang, Radio Sci. 4(12) (1962)].

TUESDAY MORNING, 27 NOVEMBER 1979

GRAND BALLROOM I, 8:30 TO 11:50 A.M.

## Session D. Psychological Acoustics I: Detection, Frequency Analysis, and Temporal Analysis

Craig C. Wier, Chairman

*Department of Speech and Hearing Sciences, University of Washington, Seattle, Washington 98195*

Chairman's Introduction—8:30

### Contributed Papers

8:35

**D1. Internal noise measurements.** Murray Spiegel and David M. Green (Laboratory of Psychophysics, Harvard University, Cambridge, MA 02138)

The purpose of this study is to compare two ways of estimating the ratio of internal noise variability to external noise variability ( $\sigma_i/\sigma_e$ ). One method [D. M. Green, Psychol. Rev. 71, 1392–1407 (1964)] infers internal noise from the consistency a listener shows in choosing between two different noise maskers when the same pair are repeated on another trial. The second method [R. A. Siegel, masters thesis, M.I.T. (January 1979)] infers internal noise from the difference in signal detectability between trials containing a pair of identical versus different maskers. The masker bandwidth (narrow = 100 Hz or wide = 2800 Hz) and mode of presentation (continuous,

burst, or continuous with gaps surrounding the observation intervals) were varied. The observation interval was 10 ms. Both methods estimate the ratio ( $\sigma_i/\sigma_e$ ) to be within experimental error of unity for most conditions. Factors that influence the estimate will be discussed, such as temporal uncertainty, response bias, binomial variability in the estimate of detection performance, and incoherence between signal and masker. [Work supported in part by a grant from the National Institutes of Health, Public Health Service, U.S. Department of Health, Education and Welfare.]

8:50

**D2. The ear's two temporal integrators, operating in series.** M. J. Penner (Department of Psychology, University of Maryland, College Park, MD 20742)

The threshold of a click is traced in the presence of two equivalent noise maskers as a function of the time  $\Delta T$  between the maskers. Maskers are said to be equivalent if any one of them alone produces the same amount of masking of a click. The time between the offset of the first masker and the click was fixed at 9.25 ms. For very brief  $\Delta T$  ( $\leq 2$  ms) the click threshold is constant, exceeding the single masker threshold by 3 dB. As  $\Delta T$  increases from 2 to 3 ms, the click threshold rises abruptly. For  $\Delta T > 3$  ms the threshold is again constant, exceeding the single masker threshold by about 9 dB. These data are interpreted as support for an early temporal integration system that summates auditory input for 3 ms or less. The output of this early system is modeled as the input to another system that applies a compressive nonlinear transformation and summates auditory input for a considerably longer period of time.

9:05

**D3. Temporal integration and the microstructure of the detection threshold curve.** Marion F. Cohen (Department of Speech, University of Connecticut, Storrs, CT 06268)

The detectability of sinusoids as a function of duration was measured for tones having durations of 10, 20, 40, 80, 160, and 320 ms and rise/fall times of 4 or 10 ms. Frequencies were selected to correspond to the peaks and valleys of each subject's detection threshold curve between 1000 and 1400 Hz [Cohen, J. Acoust. Soc. Am. Suppl. 1 65, S55(A) (1979)]. Results indicate that the threshold elevation which accompanies shorter duration is substantially larger for tones at those frequencies having a lower detection threshold level for a 350-ms tone (valleys) than it is for tones at those frequencies having a higher detection threshold level (peaks). This is explained in terms of the spread of energy which accompanies the shortening of duration. This energy spread would be expected to differentially affect detection threshold levels for peaks and for valleys. [Work supported by the University of Connecticut Research Foundation.]

9:20

**D4. Perceptual dependency of pitch and duration in categorization of pure tones.** Blas Espinoza-Varas (NSMRL, Auditory Department, Submarine Base, Groton, CT 06340)

To explore possible auditory bases of perceptual categorization, 62 naive listeners were asked to assign tones into one of four classes: "low (pitch)-short (duration)," "low-long," "high-short," and "high-long." The stimuli were sets of 81 pure tones obtained by factorial combination of nine values of frequency with nine values of duration. In one condition, frequency ranged from 0.25 to 1.0 kHz in  $\approx 100$ -Hz steps, and duration from 100 to 180 ms in 10-ms steps; in another condition the ranges were 0.536 to 1.0 kHz in 58-Hz steps and 100 to 356 ms in 32 ms steps, respectively. The tones were presented at a rate of 1/3 s and the SPL was 60 dB. A single stimulus, absolute judgement task was used and only the very initial performance was studied (first 10 min). The 50% "low-high" pitch boundary was measured as a function of tonal duration, and the 50% "short-long" duration boundary measured as a function of tonal frequency. Listeners were able to consistently categorize the tones into the four classes from the first presentation of the set of tones. For both stimulus conditions, the pitch boundary was placed around the arithmetic mean of the frequency range ( $\bar{X}_f = 0.61$  or 0.77 kHz) irrespective of tone duration. The duration boundary was generally placed below the arithmetic mean of the duration range  $\bar{X}_d$  for frequencies below  $\bar{X}_f$ , but above  $\bar{X}_d$  for frequencies higher than  $\bar{X}_f$ . That is, the higher the pitch the shorter the perceived duration. The results suggest that in auditory categorization with naive listeners, pitch can be judged independently of duration, but the duration judgement is not independent of pitch. [Supported by NRC Postdoctoral Fellowship to the author.]

9:35

**D5. Discrimination among two-tone complexes differing in frequency separation.** Søren Buus (413 MU, Department of Psychology, Northeastern University, Boston, MA 02115)

The just noticeable increase in bandwidth of a two-tone complex was measured as a function of bandwidth at three center frequencies: 500, 2000, and 4000 Hz. The bandwidths of the stimuli ranged from one-fourth to four times the critical bandwidth, the overall intensity was 60 dB SPL and the duration of each two-tone complex was 600 ms with 25 ms rise-fall time and 600 ms interstimulus interval. Using a same-different paradigm, discrimination threshold was determined with a transformed up-down procedure. The just noticeable increase in bandwidth is small at narrow bandwidths, but increases dramatically once the bandwidth exceeds some limit. At 500 and 2000 Hz, the limit corresponds to the critical bandwidth, whereas at 4000 Hz, the limit corresponds to approximately half the critical bandwidth. As the bandwidth is increased beyond two to three times the critical bandwidth, a second change in the function occurs, and the just detectable increase in bandwidth remains constant or decreases slightly. These results suggest that the interaction between two tones within a critical band can give rise to a new, discriminable cue, i.e., envelope frequency. [Work supported by NIH Grant No. 2R01NS07270-07A1 and HEW Grant No. RR07143.]

9:50

**D6. Masking of short probe sounds by tone bursts with a sweeping frequency.** F. Coninx and G. F. Smoorenburg (Institute of Perception TNO, Soesterberg, The Netherlands)

The masked threshold of a short probe tone or a short one-third octave probe noise appears to increase if the frequency of a tonal masker is swept. Frequency sweeps were exponential (octave/s) and unidirectional. Probe sounds were presented in the time center of the masker at the center frequency of the masker. The sweep gradient appeared to be an important parameter, masker duration was much less important. For 10-ms probes, 100-ms maskers, and upward sweeps increase of the masked threshold appeared to be maximal at a sweep gradient of 30 octave/s and the masked threshold was 21 dB higher than the masked threshold found for the steady masker (0 octave/s). Above 30 octave/s the masked threshold decreased. For downward sweeps masking was maximal at 20 octave/s and the threshold was 15 dB higher. The results are inconsistent with current models of masking: sweeping the frequency, the masked threshold increases whereas the energy within the critical band at the probe frequency decreases. [Research supported by the Netherlands Organization for the Advancement of Pure Research, ZWO.]

10:05

**D7. Effects of temporary threshold shift on combination-tone generation and two-tone suppression.** G. F. Smoorenburg (Institute for Perception TNO, Soesterberg, The Netherlands)

In 1972 (G. F. Smoorenburg, J. Acoust. Soc. Am. 52, 615-632) I reported that the combination tone  $2f_1 - f_2$  (CT) is not generated in the ear if  $f_2(f_2 > f_1)$  is inaudible because of a permanent threshold shift (PTS) whereas the same stimulus does generate the CT in normal ears. The PTS was local such that no PTS was present in  $2f_1 - f_2$ . Recently, a similar experiment was performed with a local temporary threshold shift (TTS). TTS was induced by a 2000 Hz, 110 dB SL, 5-min stimulus. During the measurements TTS dropped from 30 to 10 dB in the  $f_1, f_2$  region, there was no TTS at  $2f_1 - f_2$  (=1800 Hz,  $f_2/f_1 = 1.15$ ). For  $L_1, L_2 \leq 40$  dB SL (re normal threshold) a marked decrease of CT level was found after inducement of TTS. Up to 40 dB SL suppression effects are small and TTS effects were not yet found (suppressor at  $f_1$ , supressee at  $2f_1 - f_2$ ). For stimulus levels above 40 dB SL there was little or no effect on both CT-generation and

Two-tone suppression. The results show that the nonlinearity generating the CT is affected by reversible, presumably sensory-neural, changes and that it is not simply an aspect of basilar membrane mechanics.

10:20

**D8. The effects of high-frequency masking bands on midfrequency frequency discrimination.** M. E. Stanton and D. A. Nelson (Hearing Research Laboratory, Departments of Otolaryngology and Communication Disorders, University of Minnesota, 2630 University Avenue S.E., Minneapolis, MN 55414)

To evaluate the contribution of high-frequency hearing during frequency discrimination at lower frequencies, an adaptive 4-IFC pure-tone frequency discrimination task was performed by two highly practiced normal-hearing listeners at signal levels from 10 to 80 dB SPL, with and without high-frequency band-pass masking noises. The spectra of the high-frequency masking noises ranged from 1.8 to 2.1 kHz, from 2.8 to 3.1 kHz, and from 4 to 8 kHz. The DLF-intensity functions, describing DLFs as a function of sensation level, that were obtained in the presence of the high-frequency masking noises were essentially the same as the DLF-intensity functions obtained without masking noises. DLF-intensity functions obtained from one of the two listeners were in close agreement with the results reported by Wier *et al.* [J. Acoust. Soc. Am. **61**, 178–184 (1977)]. The functions from the other listener were similar in form, but exhibited consistently larger DLFs at all sensation levels. No practice effects were found throughout the investigation. [This research supported by NIH research grant NS12125.]

10:35

**D9. Psychometric functions in pure-tone masking.** Gregory P. Widin, Neal F. Viemeister, and Gregory H. Wakefield (Department of Psychology, University of Minnesota, Minneapolis, MN 55455)

Psychometric functions were obtained for detection of a 20-ms, 1-kHz sinusoid, in the presence of forward or simultaneous pure-tone maskers. The psychometric functions are considerably steeper in simultaneous than forward masking. For example, the range of the psychometric function for on-frequency masking between the 60% and 90% correct points is 3–4 dB for simultaneous masking and about 10 dB for forward masking. The slope of the functions appears to vary little as masker frequency is changed, for constant probe frequency, in either simultaneous or forward masking. Thus, comparisons of selectivity in simultaneous and forward masking will not depend on performance level. The steep slope in the simultaneous-masking condition is consistent with earlier observations. [L. A. Jeffress, J. Acoust. Soc. Am. **58**, 399–403 (1975); D. M. Green and J. Nachmias, J. Acoust. Soc. Am. Suppl. **1** **65**, S59(A) (1979)], and may reflect relatively high uncertainty regarding signal onset and duration for short signals. [Research supported by Sigma Xi, NINCDS, NICHD, NSF, and the Graduate School of the University of Minnesota.]

10:50

**D10. Frequency selectivity. A comparison of methods.** Craig C. Wier and Susan J. Norton (Department of Speech and Hearing Sciences, University of Washington, Seattle, WA 98195)

Forward-masking and pulsation thresholds were measured using both method-of-adjustment and two-alternative, forced-choice psychophysical procedures. The forced-choice pulsation-threshold procedure was similar to that described by DuMond and Stern [J. Acoust. Soc. Am. Suppl. **1** **65**, S58(A), (1979)]. Signals and maskers were digitized sinusoids. Two sets of stimulus conditions were tested: 250-ms masker and 20-ms probe, and 125-ms masker and 125-ms

probe. These conditions approximate those often reported for forward-masked and pulsation thresholds, respectively. Data were plotted as psychophysical tuning curves for evaluation. Differences in frequency selectivity measured across stimulus conditions and psychophysical procedures are the result of a combination of factors, including stimulus-ensemble characteristics and decision criteria. These stimulus and procedure variables must be taken into account when comparing estimates from different laboratories. [Research supported by PHS Grant No. RR-07096 awarded to the University of Washington.]

11:05

**D11. Abstract withdrawn.**

11:20

**D12. Discrimination of time-varying signals.** M. J. Collins and H. Stromberg (Communication Sciences Laboratory, CUNY Graduate School—Speech and Hearing Sciences, 33 W. 42 Street, New York, NY 10036)

In general, modelling of the auditory system on the basis of long-term spectral characteristics of signals has proved adequate for prediction of monaural performance. There is evidence, however, that for time-varying signals, such as short duration tonal glides, detectability of signals with identical long-term spectra may differ due to other characteristics, e.g., direction and rate of frequency changes [Collins and Cullen, J. Acoust. Soc. Am. **63**, 469–473 (1978)]. In the present study, difference limens were determined for signals with similar long-term spectra which were different in the time-frequency domain. The stimuli were signals composed of 30 ms tonal glides preceded or followed by 200 ms fixed frequency segments. Measurements were made about reference glide segments with 0-, 17-, or 34-Hz/ms linear rate of frequency change. Experienced, normal-hearing listeners served as subjects, and an adaptive procedure was used. Smaller difference limens were obtained when glide segments followed rather than preceded the fixed frequency segment. This finding cannot be easily explained by peripheral events alone. Rising glides tended to be more discriminable than falling glides when comparing signals with glide segments of equal rate and temporal position relative to the fixed frequency segment. The latter observation can be accounted for on the basis of phase differences between rising and falling, spectrally similar signals, and a model will be presented. [Work supported by NINCDS, Grant No. 1F32 NS 06053.]

11:35

**D13. Temporal order bias in an AXB paradigm.** M. J. Collins (Communication Sciences Laboratory, CUNY Graduate School—Speech and Hearing Sciences, 33 W. 42 Street, New York, NY 10036)



**E4. Analysis and display techniques for correlating articulatory and phonetic features in speech production.** W. L. Nelson, O. Fujimura, and S. Alfredson (Bell Laboratories, Murray Hill, NJ 07974)

A large body of speech-articulation data is being assembled using the x-ray microbeam system [Fujimura, Kiritani, and Ishida, *Comp. Biol. Med.* 3, 377-381 (1973)]. Once the phonetic elements of the speech utterances have been aligned with the articulatory data [Nelson, *J. Acoust. Soc. Am. Suppl.* 1 65, S22 (A) (1979)], it is possible to access and analyze the articulatory and acoustic data by phonetic feature. This paper discusses several techniques which have been developed for both statistical and detailed analyses of articulatory-phonetic relationships. In particular, two-dimensional histograms of x-ray pellet positions for many occurrences of a phonetic feature are useful for testing various assumptions and hypotheses concerning the principal articulatory conditions for that feature. For example, the retraction of the back of the tongue characterizing /l/ in different phonetic environments [Fujimura, Miller, and Escobar, *J. Acoust. Soc. Am. Suppl.* 1 61, S48 (A) (1977)] can be easily and quantitatively demonstrated.

**E5. Kinesthetic reaction times in the speech motor system.** K. J. Cole and J. H. Abbs (Speech Motor Control Laboratories, 1500 Highland Avenue, Madison, WI 53706)

Several speech production theorists have cited long-latency RT (reaction times) in the speech musculature as a partial basis for discounting the potential role of afferent feedback. However, these RT latencies, ranging from 120 to 140 ms, have been obtained primarily with auditory and visual stimuli. In the present experiment we were interested in determining the *minimal response latency capabilities* of the speech control system under optimal learning and stimulus conditions. Subjects were instructed to produce labial closure following a brisk mechanical stretch stimulus applied to the oral angle. In most subjects, the voluntary orbicularis oris (OOI) response consisted of: (1) a marked reduction of "resting" EMG activity, followed in 10-15 ms by, (2) the interference pattern related to activation of OOI. In highly trained subjects, RT latencies from the onset of the stimulus to the onset of OOI activation were frequently as short as 50 ms with a large percentage less than 75 ms. Latencies to the onset of the EMG reduction were correspondingly shorter. These findings will be discussed in relation to (1) comparable "kinesthetic" RT results recently reported in the spinal motor system, and (2) implications for the role of afferent mechanisms for speech motor control. [Work supported by NSF grant I3274.]

**E6. Anticipatory labial coarticulation: Extent and influencing variables.** James Lubker, Robert McAllister, and Wendy Linker (Institute of Linguistics, Department of Phonetics, University of Stockholm, 106 91 Stockholm, Sweden)

This research is designed to investigate the temporal extent of anticipatory (right-to-left) coarticulation as well as the variables which may influence it. Previous research in this and in other laboratories has suggested that anticipatory coarticulation in lip rounding extends for as much as six consonants or 600 ms prior to a rounded vowel. Some recent work, however, has suggested a more limited extent of such coarticulation. The present work is an attempt to resolve this issue and thus shed some light on cognitive representation and input units for the motor control processes of speech generation. Additionally a number of variables have been proposed to influence the precise location of the anticipatory boundary of labial coarticulation; for example front versus back tongue position for the rounded vowel, stress of the rounded vowel preceding the consonant string. In the present experiment these and other variables are controlled and their effect on the onset of coarticulation is described.

**E7. A kinematic experiment of lip and jaw coordination.** Akira

Hasegawa (Biocommunication Research Laboratory, University of Alabama in Birmingham, University Station, Birmingham, AL 35294)

The objective of this study was to document important kinematic characteristics of lip and jaw coordination. A new experimental paradigm was developed, which involved matching computer-controlled, video-displayed articulatory targets to articulations with or without visual feedback. The study was conducted with three male adult subjects to examine motor equivalence for vertical lip separations of 10, 20, 30 and 40 mm. A series of "target dots" were displayed one at a time on a TV monitor in front of the subject. The target dot indicated the vertical distance between the upper and lower lips. The subject was first trained with the aid of another dot on the screen which indicated vertical lip separation between his lips as determined by lateral gnathometry. The subject was instructed to follow the target as accurately as possible. Data collection in lateral gnathometry was carried out first with visual feedback and then without visual feedback. The results indicated that target lip separations were achieved by systematically different combinations of lip and mandible displacement. [Work supported by NIH grant NS11852-04.]

**E8. The relationship between jaw length and jaw opening for vowels.** W. G. Ewan and T. Gay (Department of Oral Biology, University of Connecticut Health Center, Farmington, CT 06032)

Almost nothing is known about the physiological and acoustic aspects of speech adaptation among speakers whose oral structure is modified by trauma or surgical revision. For example, in speakers who undergo surgical shortening or lengthening of the mandible, it is not clear how the revised jaw length is related to articulatory compensation. Systematic changes in jaw position for speech due to surgical revision would provide insight into the biomechanics of normal jaw movement and speech motor control. The relation between "jaw length" and jaw opening for steady-state vowels has been investigated using both patients undergoing surgical changes in jaw length and a group of normal controls. Photographic and photoelectric transducing techniques were used to monitor jaw movement during the production of the steady-state vowels [i, e, æ, o, u, ʌ]. Patients exhibited an increase in jaw opening with surgical shortening and a decrease in jaw opening with surgical lengthening. Controls (adults and children) with a large range in jaw size had a similar pattern in that a speaker with a larger jaw had less jaw opening for the steady-state vowel [a] than a speaker with a smaller jaw. The biomechanical and phonetic factors of this inverse relationship between jaw size and jaw opening will be discussed.

**E9. Tongue activity in compensatory speech: The use of articulatory models to interpret acoustical data.** S. Hamlet and E. Reeves (Department of Hearing & Speech Sciences, University of Maryland, College Park, MD 20742)

The question addressed was whether, in compensatory speech, the vowel following a consonant articulated with an advanced or retracted tongue placement, would also show evidence of an advanced or retracted tongue posture. Palatographic data were used to determine tongue contact region for consonants. An articulatory model of tongue configuration [Harshman, Ladefoged, and Goldstein, *J. Acoust. Soc. Am.* 62, 693 (1977)] was used to interpret vowel formant frequency data for vowels. Additionally, vowel formant frequencies were adjusted to account for the effects of known compensatory jaw position. With jaw position corrections included the results were that vowels following a retracted or advanced articulation of sibilant and alveolar stop consonants, showed a similar retracted or advanced set of the tongue. This was not the case for vowels following /l/. Without adjustments for jaw position the model predictions did not indicate a similarity in tongue set for vowels and consonants for any of the CV syllable types. [Work supported by NIH Grant DE-03631.]



**E10. Modeling articulatory compensation in terms of physiological parameters.** Peter Ladefoged and James T. Wright (UCLA Phonetics Laboratory, UCLA, Los Angeles, CA 90024)

An 18-section computer model of the vocal tract was used to investigate articulatory-acoustic relations by mapping from an articulatory feature space to an acoustic formant frequency space. The articulatory space was defined in terms of a five dimensional matrix in which the dimensions represented (1) degree of raising of the front of the tongue, (2) degree of raising of the back of the tongue, (3) extent of lip aperture, (4) degree of rounding, and (5) height of the larynx. The tongue dimensions and the lip dimensions were rescaled to satisfy a definition of equal articulatory step size in terms of similar differences in the total midsagittal section of the tongue and in the square root of lip area. Examination of the formant frequency space suggested that changes in tongue shape alone produce unique sets of values for the first three formant frequencies. However, variations in tongue shape combined with variations in lip area and larynx height result in articulatory compensation giving equivalent values for the first three formants. Changes in tongue shape alone failed to generate plateau-like regions of acoustic stability. [Work supported by NIH.]

**E11. Determination of lower limits of pitch variation in normal larynges.** D. Coulter, C. Ludlow, and F. Gentges (NINCDS, NIH, Federal Building, Room 1C-11, Bethesda, MD 20205)

It was desired to set a lower limit on the required sensitivity of an  $F_0$  frequency detector to assist in establishing baseline data on normals for early larynx pathology detection. Synthetically generated vowels with  $F_0$  jittered by a small amount were used to evaluate an analog pitch detector developed for this purpose. This detector was found capable of detecting peak-peak jitter values of less than  $3 \times 10^{-6}$  s. The detector was then used to evaluate the pitch of extended vowel productions. Mean adjacent differences, mean deviations of cycles from predicted values based on adjacent cycles, and mean jitter ratios were computed. The subjects evaluated were normals whose smoking and alcoholic consumption histories were known. Their vocal outputs were simultaneously monitored with acoustic microphone, accelerometer, and laryngograph pickups. Each subject was subsequently examined for larynx normalcy. It was concluded from the results of the jitter analysis that to ensure accurate description of the jitter of normal larynges it is necessary for the detector to be sensitive to at least  $5 \times 10^{-6}$  s changes in period.

**E12. The effect of stress and task variation on formant location.** K. R. Scherer and F. Tolkmitt (Department of Psychology, University of Giessen, Giessen, West Germany)

The effect of psychological stress and of different task demands on the frequency, amplitude, and band width of formants in words spoken as part of a word recognition task was studied. Subjects were 39 medical assistants who took part in an experiment on the effect of arousal on speech performance. Subjects had to engage in two types of cognitive tasks with different demands on concentration and mathematical ability and had to describe two types of films, one film on pottery making with low stress inducing potential and one film on a surgical operation. Immediately after each of these tasks, subjects had to read out loud words which were presented at relatively short exposure times (to simulate a word recognition task). Vowels within these words were isolated and formant frequencies, amplitude and band width were measured using linear prediction formant estimation routines. The results are interpreted in terms of the inter-speaker and intra-speaker variability of formant data across different levels of emotional arousal and task demands.

**E13. Laryngeal adjustments in stutterers.** H. Yoshioka<sup>a)</sup> and A. Löfqvist<sup>b)</sup> (Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

Simultaneous recordings of voice and photoelectric glottogram during the speech production by stutterers were made. The subjects were asked to produce as quickly as possible each of 36 different meaningful words in response to a visual stimulus. The results revealed that some cases of part-word (word initial voiceless stop) repetitions were accompanied by one or two extra opening gestures corresponding to the repeated aspiration noises. Similar trial-and-error types of the glottal movement were also detected even during silent delayed response time in some other cases which were auditorily judged as fluent. As for the production of the words begun with voiced sounds, unnecessary opening gestures prior to the vocalization for the initial voiced segment were sometimes observed, regardless of the judgements on its fluency and/or on the length of its latency. These findings suggest that at least this type of stuttering is linked to temporal disruption of the precisely controlled abducting and adducting gestures of the glottis, which are indispensable to fluent speech production. [Work supported by NINCDS and BRSG.]

<sup>a)</sup> Also University of Tokyo, Tokyo, Japan.

<sup>b)</sup> Also Lund University, Lund, Sweden.

**E14. Articulation of [i]'s.** John J. Ohala and Haruko Kawasaki (Phonology Laboratory, Department of Linguistics, University of California, Berkeley, CA 94720)

The articulation of the palatal vowel [i] in various phonetic environments was studied in one speaker of English and one speaker of Japanese using the technique of dynamic contact palatography. The utterances examined were of the form  $C_1V_1C_2V_2$ , where  $V_1$  or  $V_2$  was [i][u] or [a] and the other vowel always [i].  $C_2$  was any of a variety of consonants proper to the languages involved. Preliminary results suggest that the articulation of [i] is influenced in complex ways by both consonantal and vocalic environment. For example, "front" consonants such as [n] serve to preserve the canonical contact pattern for [i] whereas "back" consonants such as [k] permit greater deviation from the canonical form due to coarticulation with adjacent vowels. Other things being equal, [i] seems to dominate in the coarticulation between vowels in the speech of the Japanese speaker. A momentary reduction in the extent of palatal contact during the consonant in an [ipi] utterance was noted. This is apparently due to passive enlargement of the oral cavity caused by impinging oral pressure. [Work supported by NSF.]

**E15. Effectiveness of glossectomee palatal augmentation as measured by dynamic palatometry and analysis-by-synthesis techniques.** J. M. Christensen and A. Hasegawa (Department of Biocommunication, University of Alabama, Birmingham, AL 35294) and J. E. Hutton (Department of Maxillofacial Prosthetics, University of Alabama, Birmingham, AL 35294)

Dynamic palatometry and analysis-by-synthesis techniques were used to determine effectiveness of palatal vault augmentation to enhance the speech capabilities of a glossectomee. The fricatives /s/ and /ʃ/ were symmetrically combined with three vowels to form six nonsense words which were repeated in a randomized order within the frame sentence. "Say /hACVC/ again." During three evaluation sessions, palatometric and acoustic measures were processed using a PDP 11/40 computer system. In the three assessment sessions, the glossectomee used (1) an unaugmented prosthesis for baseline data, (2) an augmented palate in a A-B-A design, and (3) the augmented prosthesis after having been worn for four weeks. The initial baseline data was essentially the same for /s/-/ʃ/ comparisons. The augmented pseudopalate data indicated a difference in minimal groove size, but not in location for /s/ vs /ʃ/. Acoustically, the /s/ was similar to /ʃ/. These results suggest that the cavity anterior to the groove plays a critical role in distinguishing between these two fricatives. [Work supported by NIH grant NS11852-03.]

**E16. New computer-based speech-training aid for the deaf.** Richard P. Lippmann and Charles S. Watson (Boys Town Institute for Communication Disorders in Children, Omaha, NE 68131)

A computer-based speech-training aid, under development, employs the form of template analysis and comparison used in current commercial systems for the recognition of isolated words or short phrases. In this aid, as in such systems, a talker's own utterances are acoustically analyzed to form reference templates to which later utterances are compared. In one training mode, the aid will identify which of a stored list of words has been spoken, and demonstrate to the deaf speaker that utterances which are produced reliably are recognizable, whether or not they are intelligible. This same mode can be used to give a child control over gamelike devices, to promote a high rate of self-paced oral output. In a second, and more novel, training mode, the child's previous most intelligible speech tokens will be used as reference templates in "successive-approximation" procedures for improving the intelligibility of important words or phrases. The use of reference templates selected by perceptual criteria makes it unnecessary that this aid be selectively sensitive to the acoustic correlates that characterize deaf speech. Preliminary data demonstrate the general feasibility of this proposed aid. [Work supported by NIH.]

**E17. Acoustic characteristics of correctly produced vowels in deaf children's speech.** M. J. Osberger, H. Levitt, and R. Slosberg (Ph.D. Program in Speech and Hearing Sciences, City University of New York, New York, NY 10036)

An analysis was performed on the acoustic characteristics of the vowels /i, æ, u/ in several CV contexts in the speech of deaf and normal hearing children. The CV syllables were embedded in sentences, and only those syllables judged by a group of phoneticians to have been correctly produced were analyzed for the deaf children. For all three vowels, there was greater variability among the deaf subjects than among the normal hearing subjects in the frequency location of the first two formants. The average differences in the  $F_1 - F_2$  plots between the two groups of speakers varied systematically depending upon the vowel produced and the phonetic context in which it occurred. Differences between the two groups of speakers was observed for the  $F_1/F_2$  ratio as well as for the absolute frequency location of the formants. These data indicate that vowels in deaf children's speech can be correctly identified even though the absolute and relative formant values differ from that of normal hearing speakers. [Research supported by PHS Grant #09252 from NINCDS.]

**E18. Long-term average speech spectra for normal and hearing-impaired adolescents.** C. Formby and R. B. Mosen (Central Institute for the Deaf, 818 S. Euclid, St. Louis, MO 63110)

Acoustical aspects of the speech of hearing-impaired persons are typically examined over brief periods of time for simple utterances. We were interested in whether abnormal acoustical patterns would be observed when speech was examined over relatively longer periods of time. Speech spectra were averaged for hearing-impaired and normal adolescents using a constant frequency analysis bandwidth of 60 Hz across a 10-kHz range (integration time approximately 30 s). Comparison of the spectra reveals: (1) hearing-impaired adolescents with highly intelligible speech and normal adolescents exhibit similar spectra; (2) hearing-impaired adolescents with speech intelligibility judged to be no higher than 72% yield spectra which are sometimes different from that of the normal adolescent; (3) the most obvious of the abnormal patterns is (a) the general finding of irregularity of the spectra; (b) a reduction in acoustical power between the fundamental speech frequency and the fourth harmonic which is almost twice that seen in normal spectra; and (c) the observation that for three of the 11 least-intelligible adolescents, the third harmonic of the fundamental speech frequency is of greater amplitude than is the second harmonic.

**E19. Acoustic correlates of vocal tension/harshness in the speech of the deaf.** Robert Whitehead and John Bancroft (National Technical Institute for the Deaf, One Lomb Drive, Rochester, NY 14623)

Audio recordings were made of twenty young-adult hearing-impaired males while reading the first two sentences of the Rainbow Passage. The subjects exhibited a range of severity of vocal tension/harshness as judged from the audio recordings by a panel of three trained listeners. The first ten seconds of the continuous speech samples were digitized and analyzed using an autocorrelation computer program that quantified the periodicity of voiced speech. This measurement compared the power between the periodic portion of the voice (the signal) and the uncorrelated aperiodic component of the wave (the noise) and was expressed as a signal-to-noise ratio [J. C. Bancroft, J. Acoust. Soc. Am. Suppl. 1 65, S67(A) (1979)]. The computer program also extracted the average fundamental frequency ( $F_0$ ) for the voiced portion of the signal, as well as the amount of variability of  $F_0$ . Significant correlations were obtained between the perceptual judgments of severity of vocal tension/harshness and: the signal-to-noise ratios; and, the amount of variability in  $F_0$ . [Work supported by HEW.]

**E20. Effects of vibrotactile feedback upon the production of co-articulated CV syllables.** F. A. Saunders (Smith-Kettlewell Institute, 2232 Webster, San Francisco, CA 94115) and B. Franklin (Department of Special Education, San Francisco State University, San Francisco, CA 94132)

Eight profoundly deaf and deaf/blind children are receiving ongoing training in the production of speech sounds. An electrocutaneous sensory aid, worn around the abdomen, displays the speech sounds of the teacher and the child as vibrotactile patterns. In the first training activity, the syllables /ba (ba)/, /bee (bi)/ and /boo (bu)/ were presented (1) as a 2-s sustained utterance (to assess the child's ability to control articulators in sustaining a vowel), and (2) as a three-syllable utterance, repeated at a 3/s rate (approximating the rate of normal speech). Training stimuli were presented by videotape, and the children's responses were recorded and spectrographically analyzed, over a training period of three weeks. An analysis was made of the suprasegmental features of pitch and rhythm, as well as the articulatory features of transition and formant structure. The children's performance was compared under two conditions, one while wearing their hearing aids plus the vibrotactile belt, the other with hearing aids alone. [Work supported by NIH.]

**E21. Acoustic characteristics of the /I-i/ confusion in deaf children's speech.** H. Levitt, M. J. Osberger, and H. Stromberg (Ph.D. Program in Speech and Hearing Sciences, City University of New York, New York, NY 10036)

The purpose of this paper is to report preliminary data on the acoustic characteristics of error patterns in deaf children's speech. The /I-i/ confusion was analyzed first because of its high weighting with respect to reduction in percent intelligibility on a multiple linear regression analysis. An acoustical analysis was performed on /i/ in CV syllables which were produced in a sentence context by normal hearing and deaf speakers. Based on the transcriptions of a group of phoneticians, the deaf speakers were then divided into two groups: (1) those who correctly produced the /i/ and (2) those who substituted /I/ for the intended vowel /i/. The data revealed systematic differences in formant values for the three groups of speakers. The two groups of deaf speakers showed the greatest deviation from the normal group in the frequency location of  $F_2$ . Speakers who produced the /I/ for the /i/ showed the greatest deviation from normal. Measurement of vowel duration revealed no differences between the mean vowel duration for the two groups of deaf speakers, although there were some individual variations. These data indicate that spectral differences associated with the /I-i/ confusion show a more consistent pattern of deviation from normal than do durational differences. [Research supported by PHS grant 09252 from NINCDS.]

## Session F. Underwater Acoustics I: Signal Processing

Lloyd D. Hampton, Chairman

*Applied Research Labs, University of Texas, P.O. Box 8029, Austin, Texas 78712*

Chairman's Introduction—9:00

*Contributed Papers*

9:05

**F1. Adaptive focusing: The likelihood ratio.** F. M. Labianca and S. P. Lloyd (Bell Telephone Laboratories, Inc., Whippany, NJ 07981)

A prerequisite for the implementation of the adaptive focusing detection scheme [A. J. Claus and F. M. Labianca, abstract F2, this session] is a temporally stable signal field distribution across the array aperture. Stability is characterized by a low-rank signal covariance matrix. The case of absolute stability, that is, unity rank, occurs rarely in practice, but is of interest because its theoretical treatment can be carried out in closed form and lends insight to the choice of practical test statistics in less stable cases. The model is constructed under the assumption that linear filtering of the acoustic array data produces a vector time series in  $p$ -dimensional unitary space  $U$ , where  $p$  is significantly smaller than the array dimension. Under  $H_0$  (noise only) the process in  $U$  is centered isotropic Gaussian, white of unit power. Under  $H_1$  (noise plus signal) the signal has constant but unknown ray direction in  $U$ , and the amplitude process along this ray is white (complex) Gaussian of specified power. The joint likelihood ratio for  $L$  observations in  $U$  is found in closed form, suitable for numerical calculation; it is a function of the eigenvalues of the  $p \times p$  sample covariance matrix. The results of Monte Carlo simulation of the model, including cumulative distributions, ROC curves and performance curves are presented with emphasis on application to cases which might occur in practice. [This work has been supported in part by the Independent Research and Development Program of the Department of Defense and in part by the Naval Electronics Systems Command, Code ELEX-320.]

9:20

**F2. Adaptive focusing: A spatial distortion-adaptive receiver.** A. J. Claus and F. M. Labianca (Bell Telephone Laboratories, Inc., Whippany, NJ 07981)

Adaptive focusing is an array-processing scheme in which the receiver is designed for adaptive detection of localized sources under conditions where the signal field distribution across the array aperture is distorted from a plane-wave distribution, but is unknown *a priori*. The mechanisms responsible for the distortion, whether simple multipath or more complex propagation conditions, do not play important roles in the scheme. Rather, the emphasis is on estimation of the complex sensor responses using frequency domain processing and subsequent use of the resulting response vector to steer the array and thus maximize its output power for the signal sector of interest. The procedure exploits the spatial structure of the source field as it manifests itself in the eigenstructure of the sample covariance matrix. Whatever the physical mechanism causing the signal field distortion, the eigenstructure is generally such that estimation can be carried out in a unitary space of dimension significantly smaller than the number of sensors in the array. Clearly, temporal stability of the field distribution and signal-to-noise ratio are important parameters. Their influence on the detector has been studied and numerical results are presented. Monte Carlo simulation

has shown that for threshold signals and under conditions for which a conventional beamformer is degraded by as much as 5 dB, adaptive focusing can perform within a fraction of 1 dB of an ideal beamformer for which the signal field was known *a priori*. [This work has been supported by the Naval Electronics Systems Command, Code ELEX-320.]

9:35

**F3. Composite signals for simultaneous estimation of target range and bearing.** D. P. Skinner (Code 791, Naval Coastal Systems Center, Panama City, FL 32407)

In a previous paper [J. Acoust. Soc. Am. Suppl. 1 65, S62(A) (1979)] the design of large time-bandwidth product signals for simultaneous estimation of the range and bearing of stationary point targets from a moving platform was discussed. Examination of the range-bearing ambiguity functions produced by a number of simple FM signals indicated marginal gains in terms of interference suppression even for relatively large TW products. In this paper the performance of a class of composite (multi-carrier) FM signals which have desirable ambiguity, and interference suppression properties will be discussed and compared with the previous results. [Work supported by NAVMAT.]

9:50

**F4. Coherence degradation in active sonar systems.** R. C. Higgins (Weapon Systems Department, Naval Underwater Systems Center, Newport, RI 02840)

This paper quantifies the degradation in performance of a quadrature receiver when the phase of the input target signal is time-varying and the background noise is additive Gaussian. The quadrature receiver, a conventional signal-processor in sonar applications, consists of a replica-correlator followed by a quadratic detector. In active sonar systems with narrowband transmit signals, direct path and multipath returns from a distributed highlight target add at the receiver to produce a composite echo which has a time-varying phase over the signal duration. As a result, a mismatch in phase exists between the received signal and the replica (replica-correlator) causing a degradation in processor performance. This paper measures the amount of processor degradation using a narrowband target signal model with Rayleigh distributed amplitude fluctuations and uniformly distributed phase fluctuations. First, an expression for the output signal-to-noise ratio of the quadrature receiver is derived using the postulated signal model. Second, the receiver's degradation is measured by defining a normalized signal-to-noise ratio and examining this parameter for the following three variations in the basic signal model: (1) the amplitude of the in-phase and quadrature signal components of the signal is constant, (i.e., complete correlation), (2) the amplitude of the in-phase and quadrature components of the signal is Gauss-Markov process (i.e., exponential autocorrelation function) and (3) the amplitude of the in-phase and quadrature components of the signal is a Gauss-

peaked process (i.e., exponential-cosine autocorrelation function). The results for each case are presented functionally and graphically.

10:05

**F5. A measurement system for coherent analysis of target scattering data.** A. D. Matthews and J. C. Neland (Naval Coastal Systems Center, Panama City, FL 32407)

Analysis of acoustic scattering from submerged elastic targets of finite dimensions can be complicated by aspect dependence. A method is being developed for measuring and combining scattering data from adjacent frequency bands and aspect angles into a single phase-coherent transfer function. Detailed spatial analysis of a target response should be possible by means of synthetic aperture processing techniques. This will permit high resolution windowing of small parts of the target. These techniques require that measurements over different frequency bands and aspect angles be phase coherent. This necessitates high precision positioning equipment and an accurate computer controlled data acquisition system. Such a system is described, and calibration procedures and results are illustrated. The method for removal of transmit and receive characteristic of transducers and electronic hardware is discussed.

10:20

**F6. Acoustic signal flutter reduction using digital time base correction.** D. C. Robinson (Bancor Corporation, 1121 San Antonio Road, Palo Alto, CA 94303)

The flutter sidebands induced by magnetic tape recorders can degrade the resolution of recorded acoustic signals to an unacceptable level or obscure low level spectral details. These flutter effects are effectively suppressed using newly developed digital time base correctors. These all electronic, stand-alone instruments reduce the signal time base error from high performance laboratory recorders ten-fold. Since the corrected signal time base error is independent of recorder performance, greater relative improvement is obtained from portable or severe environment recorders. Spectral plots of simulated Anti-Submarine Warfare acoustic hydrophone data show a 50-dB reduction of flutter sidebands and suppression of the baseband flutter spectrum to below 80 dB down from peak signal. Comparing other flutter correction techniques—Tape Speed Servo Control typically suppresses flutter sidebands 20 dB; and Subtractive Flutter Compensation only suppresses the base band flutter spectrum but has no effect on flutter sidebands. Digital time base correctors interface with any standard instrumentation recorder, providing a capability to recover recorded acoustic data with minimal loss of resolution or low level spectral detail.

10:35

**F7. The signal to noise gain of sparse array processors.** Martin J. Earwicker (Admiralty Underwater Weapons Establishment, HM Naval Base, Portland, Dorset DT5 2JS, United Kingdom)

In this paper general expressions are derived for the signal to noise gain of sparse array processors which take account of both broadband signal and noise fields and the finite averaging times of real processors. This is achieved by considering the covariance between the fluctuations in pairs of estimates of cross correlation functions between hydrophone outputs. In the case of sparse arrays comprising many hydrophones and for low input signal to noise ratios, it is shown that the signal to noise gain of such a processor is 4.3 dB less than that of a filled array "sum square" processor using the same number of hydrophones.

10:50

**F8. A binary detection model for at-sea sonar prediction.** J. E. Franklin (MAR, Incorporated, 1335 Rockville Pike, Rockville, MD 20852) and R. J. Urlick (The Catholic University of America, Washington, DC 20064)

The output of a sonar system is influenced to a large measure by the presence of fluctuating signals and noise. Because of these fluctuations, which frequently are greater than 30 dB over a period of 30 min, the prediction of the sonar output is a difficult task. This paper describes a new approach to the process of predicting whether or not there will be a detection at some time in the future. The output of a sonar is commonly clipped and thresholded to produce a binary-valued time series. This time series may be assumed to have a value of plus one, corresponding to a detection, or a value of negative one, corresponding to no detection. In this paper, a model of such a binary time series is developed from existing time series model theory in order to predict future values of the series. The binary prediction technique developed is applied to the prediction of the beam outputs of towed line-array sonars, as well as to the outputs of sonobuoys. Predictions based on the binary model give good comparison with the data, with a predictive accuracy of as much as 93% for shorter forecast lead times. Various other statistical properties of the time series are addressed in this paper.

11:05

**F9. Harmonic masking by two data windows.** John C. Burgess (Department of Mechanical Engineering, University of Hawaii, 2540 Dole Street, Honolulu, HI 96822)

Two practices sometimes used in digital spectrum analysis are (a) using a 1/10 cosine taper at each end of a data record and (b) augmenting a short data record with zeros. Each practice is equivalent to using a data window. The spectra of both a 1/10 cosine taper data window and of a rectangular data window with 10% contiguous zeros have significant side lobe structures at amplitude ratios below about -20 dB. The use of either procedure can lead to effective masking of low-level harmonic components in the spectral representation of a data record. When there are sufficient data to provide a required record length, leakage is most effectively reduced by using a data window designed for that purpose [see, e.g., J. C. Burgess, *J. Acoust. Soc. Am. Suppl. 1* 65, S61(A) (1979)]. If a data record must be augmented with zeros, the undesirable effects of the rectangular window can be reduced considerably by both using a low-leakage window and placing the zeros equally at each end of the data record.

11:20

**F10. Submarine target strength—a random process?** Louis C. Maples (Naval Underwater Systems Center, New London, CT 06320)

In an effort to determine a simple but meaningful basis for comparison of two submarine targets of the same class, it has been found helpful to utilize the arithmetic mean target strength, averaged over all aspects, as a measure. If we look at target strength as a random variable, we may plot a "density" function and examine its characteristics, possibly classify the distribution, and determine whether the use of any statistical measure other than the mean is either justifiable or useful—or do the target strengths representing different aspects also really represent different populations, different distributions? Several sets of data are analyzed in this context and the question of a realistic random aspect target strength as a possible output is considered as well.

**Session G. Engineering Acoustics II, Physiological Acoustics I, and  
Psychological Acoustics II: Hearing Aid Technology**

Sam Lybarger, Chairman

*101 Oakwood Road, McMurray, Pennsylvania 15317*

**Chairman's Introduction—2:00**

***Invited Papers***

**2:05**

**G1. Acoustic diffraction and resonance effects for the aided versus the unaided listener.** George F. Kuhn (Vibrasound Research Corp., 4673 South Zenobia Street, Bowmar Heights, Denver, CO 80236)

The acoustic pressure at the hearing aid microphone differs from the incident pressure field, be it partially or totally progressive or diffuse, primarily as a result of diffraction by the head and/or torso of the aided listener. However, for the normally hearing subject the incident pressure field, before reaching the eardrum, is additionally transformed by the pinna and the ear canal, which together produce additional resonance amplification and directional effects; neither of these effects is produced at the hearing aid microphone. Results showing the separate effects of the head, torso, and external ear on the pressure transformations of acoustic waves for incidence in the horizontal—and median vertical—planes, as well as for random incidence, will be presented. The frequency response, the horizontal—and median vertical—plane directivities and the available localization cues for aided and unaided listeners will be compared. Some schemes to restore the localization cues for the aided listener to those of the unaided listener will be discussed. Furthermore, a physical mechanism which may cause speech in noisy environments to be less intelligible for aided listeners than for unaided listeners will be proposed.

**2:35**

**G2. Subminiature transducers and their acoustic coupling: State of the art.** Mead C. Killion (Industrial Research Products, Inc., a Knowles Company, 321 Bond Street, Elk Grove Village, IL 60007)

Until recently, the transducers that were available to the hearing aid designer imposed major limitations on the performance which could be offered in headworn hearing aids. Within the last decade, however, these limitations have been largely removed. Today's subminiature microphones and earphones permit a wide variety of hearing aid characteristics, up to and including true high-fidelity performance: a claim substantiated in recent physical and psychophysical experiments [Killion, J. Acoust. Soc. Am. Suppl. 1 64, S103(A) (1978)]. The usable performance of subminiature transducers may be largely determined by their acoustic couplings—so much so that a meaningful definition of their frequency response requires that the coupling system be specified—and a crucial factor in recent performance improvements has been a systematic examination of the acoustic coupling systems available to headworn hearing aids. While sometimes considered a nuisance or worse, the acoustic couplings can be exploited in hearing aid design to (a) improve directional performance, (b) provide desirable frequency-response tailoring, and (c) reduce battery drain. Earmold construction in particular becomes increasingly important as the bandwidth of hearing aids is increased. At low (below 1 kHz), medium (1–3 kHz), and high (above 3 kHz) frequencies, the use of earmold venting, damping, and a stepped-diameter sound channel, respectively, can provide substantial (and predictable) modifications to the real-ear response of hearing aids; modifications which can often be performed at the dispenser level. Examples of the performance allowed by today's transducers and acoustic couplings will be given.

**3:05**

**G3. Choosing the frequency responses of hearing aids—the viewpoint of a communications engineer.** L. D. Braida, R. L. Dugal, and N. I. Durlach (Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Many of the issues associated with the selection of frequency responses for hearing aids are similar to those encountered in the design of voice communication systems for listeners with normal hearing. The design of such systems typically utilizes the predictions of articulation theory, which incorporate the results of a wide variety of measurements of speech transmission under conditions of restricted bandwidth, additive noise, and limited dynamic range. Although a number of investigations has



**H2. Vibrational characteristics of Saron Barong metallophones in the 1893 Field Museum Gamelan.** William R. Savage (Department of Physics and Astronomy, The University of Iowa, Iowa City, IA 52242), Edward L. Kottick (School of Music, The University of Iowa, Iowa City, IA 52242), and Sue Carol DeVale (Asian Archaeology and Ethnology, Field Museum of Natural History, Chicago, IL 60695)

The gongs used in Asiatic music frequently exhibit noticeable glides in pitch comprising a significant fraction of a semitone. Bar-form metallophones of the 1893 Gamelan also show a glide in pitch. This contrasts with western bar-form metallophones which are generally pitch stable. Measurements of the keys of two Saron Barong show that they have different lengths, widths, and cross sections but do not "scale" since they have nearly similar masses. The keys of lower pitch are longer and wider than those of higher pitch within a given seven note Saron. The upper surface of the keys is convex for both length and width aspects. The lower surface of the lower pitch keys is concave becoming convex for the upper keys. They are mounted with nails through holes at a distance one quarter the length from the end on resilient pads above individual air cavities. The holes are not symmetrically located and the bars are not straight. They were cast of bell metal in sand molds and roughly finished. The central part of the underside is hollowed. Torsional and flexural modes near the pitch frequency can be found by external excitation. These modes have frequencies consistent with those found by high-speed spectral analysis and pitch measurements. The pitch glide seems to be associated with a bar shape that favors excitation of both torsional and flexural modes at the initial strike with a more rapid decay of the torsional mode. This mechanism differs from that suggested for gongs where the moving mass associated with the mode decreases with decreasing amplitude of vibration.

2:45

**H3. A study of cymbal vibrations.** Harvey Fletcher and I. G. Bassett (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

The vibration patterns of the cymbal were studied by driving the plate at edge with a sinusoidal signal. Chladni patterns were formed for sixty of the strongest modes of vibration. Typical patterns will be shown and discussed. Analyses were made of the sound from a single cymbal struck with a mallet. The vibration is very complex with more than 600 identifiable frequencies. Analysis was done passing the recorded sound through a 3-Hz bandwidth filter to isolate single frequency components. Information on decay rates was obtained using a 10-Hz bandwidth filter.

### Contributed Papers

3:05

**H4. Vibrations of cymbals, gongs, and tamtams.** Richard W. Peterson (Department of Physics, Western Illinois University, Macomb, IL 61455) and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115).

Cymbals, tamtams, and gongs are three important examples of percussion musical instruments, which employ vibrating plates. When struck, they vibrate in many different modes of vibration and radiate sound which is rich in overtones [T. D. Rossing, *The Physics Teacher* 15, 278 (1977)]. Gongs convey a fairly strong sense of pitch, whereas cymbals have indefinite pitch. We have studied the modes of vibration and the sound spectra of a number of cymbals, gongs, and tamtams both when struck and when driven in steady state. Various techniques have been employed for studying the modes of vibration, including Chladni patterns, holographic interferometry, probing the nearfield sound, and attaching vibration sensors [T. D. Rossing and R. Ross, *J. Acoust. Soc. Am. Suppl.* 1 64, S150(A) (1978)]. In some cases the results agree closely with the simple theory of plate vibrations; in other cases, rather unusual effects are observed.

3:20

**H5. The acoustics of timpani.** Craig A. Anderson<sup>a)</sup> and Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The frequencies of the vibrational modes of kettledrum heads have long been known to differ from those of an ideal unloaded membrane

[Lord Rayleigh (J. W. Strutt), *The Theory of Sound* (Macmillan, 1894), 2nd ed., Vol. 1]. The three principal modes, under normal playing and tuning conditions, have frequencies nearly in the ratio of 4:3:2, thus giving timpani a musical pitch which is easily discernible. This favorable relationship of modal frequencies is brought about by membrane stiffness and air loading; our experiments indicate that air loading is the more important factor. We have measured the frequencies of ten or more modes of vibration under various conditions: high and low tension, with and without the kettle, in air and helium atmospheres. The kettle raises the frequency of the lowest symmetric mode, and also fine tunes some of the higher modes; its most important role is that of a baffle, however.

<sup>a)</sup> Present address: Nelson Industries, Incorporated, Stoughton, WI 53589.

3:35

**H6. Analysis of Japanese Tsuzumi tones.** R. D. Olsen and I. G. Bassett (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602)

Four basic types of tones for the (KO) Tsuzumi (an hourglass-shaped, wooden Japanese drum about 25 cm high and 10 cm in diameter with small ropes interconnecting the two heads) were analyzed using a Fast-Fourier Transform computer program. Recordings of the tone were made by Michiko Toyama 1-29-18 Hanegi, Setagaya-Ku, Tokyo, Japan. The tones analyzed are described as follows: (1) "KO," produced by striking the center of the drum head with two to four fingers of the right hand while the ropes holding the head are at first held loosely and then squeezed quickly. Twelve frequency components have been identified ranging

from 254 to 4348 Hz. Maximum intensities cover a range of 35 dB with decay rates from 80 to 597 dB/s. (2) "KU," produced in a similar manner to the KO tone, but with only one finger. Frequencies are between 311 and 3230 Hz. Intensities cover a range of 25 dB, and decay rates between 98 and 1140 dB/s. (3) "TA" produced by hitting the edge of the head with two fingers while maintaining maximum tension on the ropes. Frequencies are between 311 and 3234 Hz with little or no shift. Intensities cover a range of 38 dB, with decay rates between 96 and 1140 dB/s. (4) "CHI" produced in

the same manner as TA, but striking only with the third finger of the right hand. Frequencies of components in this recording lie between 255 and 3465 Hz. Intensities cover a 46-dB range. Decay rates are between 85 and 1355 dB/s. In both the KO and the KU tones, the most pronounced component is the lowest frequency and has the slowest decay rate. On the TA and CHI tones, however, the lowest frequency components are not the most intense. The most intense component is in the area of 780 Hz with a decay rate about four times as rapid as that of the lowest frequency component.

TUESDAY AFTERNOON, 27 NOVEMBER 1979

BONNEVILLE ROOM 5, 2:00 TO 5:05 P.M.

## Session I. Noise II: Effects of Time-Varying Noise on People

Simone L. Yaniv, Chairman

*National Engineering Laboratory, National Bureau of Standards, Washington, D.C. 20234*

Chairman's Introduction—2:00

### *Invited Papers*

2:05

**11. Noise indices for fluctuating sound levels: A review.** J. S. Bradley (Faculty of Engineering Science, University of Western Ontario, London, Canada, N6A 5B9)

A large number of noise indices have been proposed that incorporate measures of the temporal fluctuations in sound levels. Such noise indices usually relate to sound level fluctuations over periods between approximately 1 s and several minutes, but considerations can be extended to shorter periods and instrument integration times, or to longer periods and day-night differences. Unfortunately the results of field and laboratory studies have not provided strong consistent evidence to support the use of one particular index as a more successful predictor of adverse human responses to noise. It is intended to first review the various indices that have been proposed, to compare their physical similarities, and the relative difficulty of their calculation. Results from field and laboratory studies will be examined to discuss the merits of these indices as predictors of adverse human responses, and the reasons for the various successes and failures.

2:35

**12. How to measure time varying noise exposure in residential communities.** Paul N. Borsky (Columbia University Noise Unit, 367 Franklin Avenue, Franklin Square, NY 11010)

The controversy over how best to integrate time varying noise exposures in residential communities can only be resolved by an analysis of objective data secured from well-designed community noise surveys. Past community studies have provided clarification of the important complex physical and human variables involved in the harmful health and welfare effects of environmental noise, but new research is needed to quantify the number-level tradeoffs and time of day penalties. Community reactions can be compared where noise exposures are equal in day or evening but differ in the night time. The effects of ambient noise on more intense aircraft noise exposures can also be ascertained. A mail survey of the top 50 airports reveals at least 13 different time of day and type of operation situations with exposed populations up to 8–10 miles from the airport. Considering regional variation, about 16 selected airports will represent the range of physical exposures.

3:05

**13. Findings from community surveys on the noise level/number of event trading relationship.** James M. Fields (National Research Council Postdoctoral Associate, Noise Effects Branch, NASA Langley Research Center, Hampton, VA 23665)

The value of different noise indices' characteristics can be better compared with regression rather than with correlation analyses. For these community noise indices, most of the debate has centered on the way in which three characteristics of noise events should be combined: peak noise level, number of events, and duration of events. Four types of indices have been proposed to combine these variables: energy hypotheses, simplified energy hypotheses, modified energy hypotheses, and interaction effect hypotheses. By breaking down the various indices into their component variables and



considering each variable separately in regression analyses, the component parts of the theories underlying each of the indices can be directly examined. Three data sets (Tracor aircraft study, 1967 Heathrow study, and British Railway study) have been reanalyzed using this approach. Results from other studies are also reviewed. It is clear that knowledge about alternative noise indices would be enhanced in future studies if (1) component parts of the indices are separately analyzed, (2) a wide range of types of indices are compared in each research project, (3) noise level and number are not highly correlated in the sample, (4) the number of events is not confounded with other area-level variables in the sample, and (5) problems in counting numbers of events are addressed.

3:35

**14. Effect of time varying noise on speech communication.** Karl S. Pearsons (Bolt Beranek and Newman Inc., P.O. Box 633, Canoga Park, CA 91305)

A review of research on effects of time varying noise on speech intelligibility will be presented. Investigations using recorded traffic noise and shaped noise as masking will be reviewed, as well as the use of intelligibility tests. Comparisons of Articulation Index predictions will be presented for different intelligibility tests. The importance of accurate measures of speech levels will be emphasized. The relationship between speech intelligibility and annoyance will also be discussed, using results from social surveys and laboratory tests. Research needs will also be discussed.

4:05

**15. Sleep interference from intermittent and continuous noise exposure.** R. D. Horonjeff and S. R. Teffeteller (Bolt Beranek and Newman Inc., P.O. Box 633, Canoga Park, CA 91305)

Some effects of temporal patterns, duration, amplitude, and spectral content of noise intrusions on patterns of residential awakening are presented. Behavioral awakening measures developed in the course of a study of probabilities of awakening associated with low level nocturnal noise intrusions are discussed. Potential influences of background noise environments on the ability of noise intrusions to awaken people are also considered. Comparisons will also be made between the current data and those of prior studies.

4:35

**16. Effects of temporal variability of urban noise on signal detectability.** S. Fidell (Bolt Beranek and Newman, Inc., P.O. Box 633, Canoga Park, CA 91305)

A summary is provided of recent research on the effects of temporal variability of background noise on the audibility of acoustic signals. Attention will be directed to two cases: detection of intermittent signals of durations that are long with respect to short term fluctuations in noise levels; and effects of circadian variability of urban noise on detection of continuous signals of low absolute level. Detailed measurements of the variance of distributions of urban noise levels at different times of day at sites of varying population density will be reviewed, as will the practical significance of such variability for prediction of signal detectability. Means of relating annoyance from exposure to low level acoustic signals to signal detectability will also be discussed.

**TUESDAY AFTERNOON, 27 NOVEMBER 1979      BONNEVILLE ROOM 3, 2:00 TO 5:20 P.M.**

## **Session J. Physical Acoustics II: Shock Induced Effects in Condensed Media**

**Philip L. Marston, Chairman**

*Department of Physics, Washington State University, Pullman, Washington 99164*

**Chairman's Introduction—2:00**

### ***Invited Papers***

2:05

**J1. Study of higher-order elastic and piezoelectric constants with shock-wave loading techniques.** R. A. Graham (Sandia Laboratories, Albuquerque, NM 87185)

Many solids remain elastic to strains of a few percent when subjected to uniaxial strain imposed by shock loading. Nonlinear contributions are enhanced by such large elastic strains and become

accessible for relatively straightforward measurement and interpretation. Impact loading techniques have proven particularly effective for measurements of second-, third- and fourth-order longitudinal piezoelectric constants as well as third- and fourth-order longitudinal elastic constants. Shock compression studies have provided independent measurements of constants, have uniquely determined certain constants and have extended ultrasonic measurements by several orders of magnitude in strain. Typical experimental investigations and their results will be summarized and compared to corresponding ultrasonic data. [This work sponsored by the U.S. Department of Energy under Contract DE-AC04-76-DP00789 to Sandia Laboratories, a U.S. Department of Energy facility.]

2:35

**J2. Shock-wave propagation in anisotropic solids.** J. N. Johnson (MS-214, Los Alamos Scientific Laboratory, Los Alamos, NM 87545)

The systematic study of small-amplitude (peak acoustic stress  $< 1$  atm) acoustic-wave propagation in anisotropic solids has been going on for some time. The corresponding investigations of these materials with finite-amplitude impact- (or explosively) generated shock waves is still in its infancy, but is clearly becoming an interesting and useful area of scientific research. Shock-wave studies consist of single compressive stress pulses with amplitudes between  $\sim 10^4$  and  $\sim 10^5$  atm. The dynamic stress state is itself nonisotropic and can result in plastic flow as well as significant nonlinear elastic effects; in soft, ductile materials the former effect dominates the material response, while in very strong brittle materials the latter effect can be observed. This review presentation begins with the subject of shock-wave propagation in anisotropic elastic solids—common ground with classical acoustical studies—and goes on to the problem of shock-induced plastic deformation in anisotropic elastic-plastic solids. It is shown that plastic wave velocities in anisotropic materials can possess a multiplicity of values (as they do under purely elastic conditions), and that these velocities depend on the slip systems activated under dynamic conditions.

3:05

**J3. Dislocations and shock waves in LiF.** G. E. Duvall (Department of Physics, Washington State University, Pullman, WA 99164)

The shock wave produced by plane impact of a projectile on a LiF crystal develops a structure in which an initial sharp increase in pressure is followed by a stress relaxation. Pressure then passes through a minimum and a second, slower rise. The initial rise travels with elastic wave speed and is called the precursor. The second rise travels more slowly and is called the plastic wave. Precursor amplitude is found to decay as it propagates, and this decay process can be related to the generation and motion of dislocations, using the equations of micromechanics developed by J. J. Gilman. When experimental decay rates are used in this way to derive dislocation parameters, it is found that the product of dislocation density and dislocation velocity is larger by two or three orders of magnitude than can be accounted for by initial measurements of dislocation density and conventional dislocation processes. There has been some success in explaining results for Mg-doped LiF through invocation of heterogeneous nucleation of dislocations around  $\text{MgF}_2$  precipitates. But recent post shock studies do not bear out this nucleation hypothesis. In undoped LiF the decay rate is too rapid to be observed, and there is no basis for a nucleation hypothesis. The problem, its anomalies, and existing evidence will be reviewed and some speculations about unconventional processes will be aired.

3:35

**J4. Pressure-shear waves as a means for determining dynamic material properties.** R. J. Clifton (Division of Engineering, Brown University, Providence, RI 02912)

Pressure-shear waves involving changes in both normal and shear tractions on planes perpendicular to the direction of propagation are being used to study the shear strength of materials shock loaded to high pressures. The predominantly shear waves generated in these experiments are particularly sensitive to the plastic flow characteristics of the material. This talk presents an overview of theory and experiment related to pressure-shear waves. Solutions are described for boundary conditions corresponding to those of impact of two parallel plates with the direction of approach inclined to the direction of the normal of the impact plane. Experiments for generating pressure-shear waves by such impacts of skewed plates are described. Various techniques for monitoring pressure-shear waves are reviewed. Experimental results are presented for pressure-shear impact of 6061-T6 aluminum, using a transverse displacement interferometer and a normal velocity interferometer to monitor, respectively, the transverse and normal components of the motion of the rear surface of the target.

4:05

**J5. Heterogeneous yielding and melting under shock compression.** D. E. Grady (Sandia Laboratories, Albuquerque, NM 87185)

Shock compression offers a method for investigating the physical properties of solids at pressures and temperatures not readily accessible by other techniques. To correctly assess the physical phenomenon of interest, a thorough understanding of the thermodynamic and metallurgical state of the material created during shock compression is necessary. The transition to the shocked state involves a catastrophic deformation of the solid and only a limited understanding of this deformation process has been achieved. Recent studies suggest that the process of shock deformation can be heterogeneous in the sense that yielding is localized to relatively few, narrow shear zones within which the shock energy and inelastic deformation are concentrated. This concept of shock deformation can have profound consequences on the resulting shock state. Nonuniform energy deposition, strong thermal gradients, and regions of partial melt can persist for the brief period of time that the shock state is maintained, suggesting a highly nonequilibrium situation. Many of the features of shock compression and release are readily explained by a heterogeneous deformation process and apparently depend on the competing properties of thermal diffusion, melting temperature, and degree of thermal localization. [This work was supported by the U.S. Department of Energy (DOE) under contract DE-AC04-76DP00789.]

4:35

**J6. Shock-compressed liquids.** W. J. Nellis (Lawrence Livermore Laboratory, L-355, Livermore, CA 94550)

A number of liquids including  $N_2$ , Ar, and  $H_2O$  were shock compressed in the pressure range 0.1–1.0 Mbar using a two-stage, light-gas gun. Metal plates were accelerated to 7 km/s and strong shockwaves were generated on impact with targets. Equation-of-state data were obtained by measuring shock and impact velocities and applying the Rankine–Hugoniot conservation relations. Compression factors up to threefold over liquid density and temperatures up to 20 000 K were achieved. The experimental techniques will be described including the shock-impedance-matching method, the two-stage gun, the fast electronics with subnanosecond time resolution, and the cryogenic target holders. By means of the statistical mechanics calculations of Ross, the repulsive pair potential between molecules will be demonstrated to obey the principle of corresponding states for some of the liquids. Electronic excitation will be shown to have a strong influence on the shock velocity in Ar at the high temperatures achieved. [Work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore Laboratory under contract W-7405-Eng-48.]

### Contributed Paper

5:05

**J7. Experimental analysis of acoustically generated boundary cavitation.** J. A. Clark (Acousto-Optics Laboratory, Catholic University of America, Washington, DC 20064)

Cavitation can be produced by the inversion of large amplitude acoustic pressure pulses or shock waves which occurs during the reflection of waves at a soft boundary. In current experiments, acoustic pulses in water with peak pressures up to 25 atm and dura-

tions of 40  $\mu$ s and longer are being used to generate cavitation near the free liquid surface. The cavitation process is being analyzed by optical, holographic interferometry which provides whole field measurements of the density of the medium at a series of time intervals. Acoustical measurements of the echo produced by the subsequent collapse of the cavitation region are also obtained. The echo pulses are observed to be nearly as strong as the incident pulses, but are delayed by about 200  $\mu$ s relative to a small-amplitude, reflected pulse.

TUESDAY AFTERNOON, 27 NOVEMBER 1979

GRAND BALLROOM III, 2:00 TO 4:05 P.M.

## Session K. Speech Communication II: Speech Synthesis

Jared J. Wolf, Chairman

*Bolt Beranek and Newman, 50 Moulton Street, Cambridge, Massachusetts 02138*

Chairman's Introduction—2:00

### Contributed Papers

2:05

**K1. New high-frequency regeneration (HFR) techniques for voice-excited speech coders.** A. Higgins, R. Viswanathan, and W. Russell (Bolt Beranek and Newman Inc., Cambridge, MA 02138)

In voice-excited speech coders, high-frequency information must be regenerated from the transmitted baseband by some nonlinear

operation. Waveform rectification, with its well-known harmonic generation property, has traditionally been used for this purpose. Recently, Makhoul and Berouti (ICASSP '79) introduced the method of spectral folding, in which HFR is accomplished by means of aliasing the baseband spectrum. Although this method generates certain low-level tonal noises, it does not have the perceivable roughness produced by rectification and is much simpler computa-

tionally. In the present study, we have investigated several modifications to the spectral folding method with the goal of reducing or masking the tonal noises at the cost of increased computations. One modification requires randomly perturbing the nonzero samples of the spectrally folded signal, while another involves preflattening of the baseband. A third approach performs HFR in the DFT frequency domain. The paper describes these techniques and presents the results of both informal and formal speech quality evaluation of the new and existing HFR techniques. [Work sponsored by the Department of Defense.]

2:20

**K2. Lingua: A language interpreter used for demisyllable speech synthesis.** Catherine P. Browman (Bell Laboratories, Murray Hill, NJ 07974 and New York University, New York, NY 10003)

A set of programs, collectively called Lingua, have been developed to apply rules to an input string for a speech synthesis system. Lingua is a general purpose system, in which the user specifies the basic unit size for the synthesis, the set of input characters, any set of features, and any type of rule that can be expressed in linguistic notation. Lingua was used to synthesize speech from the demisyllable inventory established by Lovins [Lovins, Macchi, and Fujimura, *J. Acoust. Soc. Am. Suppl. 1* **65**, S130(A) (1979)]. The system will be described and the resulting speech demonstrated.

2:35

**K3. Application of linguistically-motivated rules of syllabication to automatic speech synthesis.** Daniel Kahn (Bell Laboratories, Murray Hill, NJ 07974)

One can derive a set of rules for the syllabication of English words and phrases using traditional linguistic argumentation—for example, simplicity of the resulting grammar, agreement with native speaker intuitions in clear cases, etc. In this paper, I describe a practical application of such rules, the automatic syllabication of phonetic strings as a component of a text-to-speech synthesis scheme whose basic unit of concatenation is the demisyllable [Fujimura, *J. Acoust. Soc. Am. Suppl. 1* **59**, S55(A) (1976)]. Some important characteristics of these linguistic rules are the use of a “maximal-initial-cluster” principle, no constraints on final clusters, and the assignment of “ambisyllabicity” in certain cases (an “ambisyllabic” consonant is one which is simultaneously the final element of one syllable and the initial element of the next syllable). Informal listening tests have lent support to the hypothesis that these linguistically motivated rules are appropriate in the practical application of syllable-based speech synthesis.

2:50

**K4. Consonant intelligibility of speech produced by dyadic rule synthesis.** Joseph P. Olive (Bell Laboratories, Murray Hill, NJ 07974) and Louis C. W. Pols (Institute for Perception TNO, Soesterberg, The Netherlands)

We measured the intelligibility of speech generated by a speech synthesis system which converts text to sound by rules. The synthesis process uses dyadic concatenation and is intended for automatic voice answerback in telephone directory assistance situations. The intelligibility of initial and final consonants in monosyllabic words was measured. For comparison two other conditions were tested: PCM-coded speech (12 bits, 10-kHz rate) and LPC-resynthesized speech (12 parameters).

The overall percentage correct score for initial and final consonants, averaged over the 33 listeners, was 93.0% and 92.8% for PCM-coded speech, 86.5% and 85.4% for LPC-resynthesized speech, and 58.2% and 73.5% for rule-synthesized speech. Each word list was preceded by two practice runs consisting of one example word

for every initial or final consonant. After this very short training phase, the subjects showed no more learning during the actual word list. The voicing character of initial voiced fricatives was not well represented in the rule-synthesis system. For most other identification errors it was clear that a more optimal choice of the dyads would cause a great improvement.

3:05

**K5. The use of LPC coefficient interpolation to improve the naturalness of sentence formation by word concatenation.** Arvin Levine (Institute for Mathematical Studies in the Social Sciences, Ventura Hall, Stanford University, Stanford, CA 94305)

A recent experiment [Sanders *et al.*, *J. Acoust. Soc. Am. Suppl. 1* **64**, S1(A) (1978)] showed that an important contributor to the unnaturalness of sentence formation by word concatenation was the use of individually recorded words (isolation forms). Experience at Stanford has shown interpolation and quantization of LPC formant frequencies and bandwidths to be effective as methods of data rate reduction. Another use for interpolation is to improve the perceived naturalness of synthesized speech by smoothing the transitions between independently recorded sounds. Several considerations for successful word-boundary interpolation are: (1) syntactic, semantic and phonetic conditions, (2) the interpolation curve—linear or more complex, (3) the durations and locations of the interpolation region in individual words, and (4) the use of a specially prepared “demisyllabic bridge” to connect words. The relative importance of smoothing transitions involving high-frequency function words (articles, auxiliary verbs, etc.) is compared to interpolating all the junctions in a sentence. [Work supported by NSF grant SED-77-09698.]

3:20

**K6. Data compression of Linear Prediction (LP) analyzed speech by interpolation and quantization of pseudo-formant parameters.** William Sanders, Carolyn Gramlich, Arvin Levine, Robert Laddaga, and Patrick Suppes (Institute for Mathematical Studies in the Social Sciences, 192 Pine Hall, Stanford University, Stanford, CA 94305)

Pseudoformants and bandwidths, derived from LP coefficients, prove highly susceptible to data compression techniques such as interpolation and quantization because of their low entropy and smooth sequential nature. We have developed a heuristic to label initially unordered pseudoformants in a way that reveals their sequential smoothness, so that linear interpolation can be successfully applied to each pseudoformant separately. A linear least-squares fit is made over several points; the number of points is limited by a bound on the allowed mean-squared fractional difference between the data and the interpolating line, and another bound on the maximum-squared fractional difference. Quantization of the resulting parameters is constrained not to introduce more error than was allowed in the interpolation procedure. All parameters need not be interpolated using the same bound; if the relative importance of the parameters is chosen in an acoustically reasonable way, interpolation can produce very high quality sound at considerably reduced bit rates. [Work supported by NSF.]

3:35

**K7. Perceived speech quality of Linear Prediction (LP) analyzed speech compressed by means of interpolation and quantization of pseudoformant parameters.** Carolyn Gramlich, William Sanders, Arvin Levine, Robert Laddaga, and Patrick Suppes (Institute for Mathematical Studies in the Social Sciences, 192 Pine Hall, Stanford University, Stanford, CA 94305)

It is possible to produce high quality LP speech with a low bit rate by interpolating pseudoformants and bandwidths derived from the

LP coefficients. To the extent that pseudoformants are like the true formants of speech, the importance of each can be inferred from classical acoustical studies. A listening experiment was performed which compared the perceived quality of speech interpolated using several assignments of importance to the pseudoformant parameters: an acoustically reasonable assignment, a uniform assignment, and a number of assignments intermediate between the two. Several levels of total distortion were used. The test shows that if an acoustically reasonable assignment of importance to each pseudoformant is given, the quality of the speech is significantly higher than if equal importance is given to each parameter. By using the acoustically reasonable allocation, very good speech quality can be maintained at a bit rate of 3200 bits/s while if the uniform allocation is used, more than 4200 bits are needed to attain the same quality. [Work supported by NSF.]

3:50

**K8. High quality system for speech transformations.** Stephanie Seneff (Room 36-513, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

A speech analysis synthesis system has been developed which is capable of independently modifying the excitation and the spectrum, and then reconstructing speech from the modified components. Since pitch extraction is not required, the reconstructed speech is more natural sounding than vocoded speech. At the core of the system is a phase vocoder. The excitation spectrum is obtained by dividing the magnitude spectrum by the vocal tract spectral model, and is then either duplicated or foreshortened to alter the number of harmonics present. Correct frequencies are restored by multiplying the unwrapped phase characteristics by the appropriate constant. The desired vocal tract shaping is then reintroduced, using interpolation between samples if necessary. The system can produce several potentially useful modifications to the voice. A male voice can be converted into a female-like voice, and vice versa. Alternatively, the spectrum can be compressed by a factor of three or more, while the pitch remains unchanged; a modification potentially useful for persons with high frequency hearing loss. A third possibility is the restoration of helium speech. In addition, the parameters of the system could be encoded to realize a voice-excited vocoder. A tape will be played illustrating the system's capabilities.

TUESDAY AFTERNOON, 27 NOVEMBER 1979

BONNEVILLE ROOM 2, 2:00 TO 4:35 P.M.

## Session L. Underwater Acoustics III: Radiation, Target Strength, Scattering, Reverberation, and Noise

Gerald J. Franz, Chairman

*Silencing Technology Associates, Box 695, Bayview, Idaho 83803*

Chairman's Introduction — 2:00

### Contributed Papers

2:05

**L1. The underwater radiated noise characteristics of a fishing trawler.** E. E. Davis and N. J. DiPiazza (Naval Ocean Research and Development Activity, NSTL Station, MS 39529)

The U.S. Coast Guard is responsible for the protection of fisheries resources within the 200 nautical mile offshore zone. In support of this requirement a study was undertaken to examine the feasibility of applying underwater acoustic monitoring techniques, for detection and classification of fishing trawler activities. The approach was to conduct a set of controlled experiments with a cooperating trawler and to employ aircraft/sonobuoy measurement techniques. The specific objectives of the experiments were to (1) determine if significantly distinctive noise characteristics are generated by the different modes of trawler operation, (2) identify the sources of any distinctive noise characteristics and (3) obtain quantitative source levels generated by the trawler for each mode of operation. The experimental objectives were accomplished and the results show a correlation of fishing operations with the radiated noise of a fishing trawler. Noise signature characteristics indicated dynamic acoustic changes during fishing operations, particularly during net deployment and retrieval. In addition, absolute trawler radiated noise levels (source levels) were determined for various fishing operations. [This work sponsored by the Commandant, United States Coast Guard, Washington, DC.]

2:20

**L2. Acoustic features of CFAV QUEST.** Arthur J. Collier (Defence Research Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B3Y 3Z7, Canada) and Ariel W. George (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138)

In 1969 the Canadian Forces Auxiliary Vessel QUEST was commissioned for use by the Defence Research Establishment Atlantic in Dartmouth, Nova Scotia. QUEST was specifically designed and constructed as a platform from which to conduct open ocean underwater acoustic experiments. The acoustic features built into the ship, some of the philosophy behind the design, and the performance of the vessel will be discussed in some detail.

2:35

**L3. Effects of source micromotion on an acoustic signal in the frequency, time, and space domains.** R. P. Flanagan and N. L. Weinberg (Institute for Acoustical Research, 615 S.W. 2 Avenue, Miami, FL 33130)

The effects of periodic source micromotion on an acoustic signal are examined with micromotion in the vertical plane being emphasized. Single phone radial and skew source motions are investigated utilizing CW and narrow-band equispaced, equal amplitude, multi-tonal signals to  $200 \pm \frac{1}{2}$  Hz. Brief comments on array reception are included. The study, based on ray model calculations, shows that source micromotion does not appreciably affect the coherence bandwidth of the received signal but introduces a fluctuation having the period of the micromotion,  $T_{mm}$ , which can reduce coherence times appreciably. Spatial coherence lengths are extremely large due to the simplified ray model. Micromotion transfers power from the central peak of the power spectrum to sidebands separated from that central peak by  $\pm 1/T_{mm}$  Hz. The power in the spectral sidebands of the received signal is dependent on the signal level and the magnitude of the source micromotion. [Work supported by ONR Code 222.]

**L4. Ultrabroadband target strength measurements of biological entities.** R. A. Johnson (Del Design, P.O. Box 1481, Silver City, NM 88061), W. A. Friedl (Naval Ocean Systems Center, P.O. Box 997, Kailua, HI 96734) and D. E. Nelson (Hydroacoustics, Inc., P.O. Box 3818, Rochester, NY 14610)

An ultrabroadband underwater echo-ranging system has been employed to collect echoes from biological entities off the coast of Southern California. The system uses an implosive impulse source and a broadband, constant beamwidth towed array. For these measurements, target strengths were calculated over the band of frequencies from 500 Hz to 50 kHz with minimal distortion due to transducer beamwidth effects. Averaged frequency spectra for echoes from many biological targets have been examined. Echoes from schools of fish exhibit Rayleigh scattering regions as well as resonances and have target strengths in excess of 0 dB with occasional schools exceeding 20-dB target strength over a portion of the band. [Work supported by the Naval Material Command.]

3:05

**L5. A numerical study of acoustic interaction effects in backscattering from linear arrays of gas-filled bubbles.** J. C. Novarini and D. R. Bruno (Department of Oceanography, Servicio de Hidrografía Naval, Montes de Oca 2124, 1271- Buenos Aires, Argentina)

The acoustic backscattering of spherical waves from a linear array of gas-filled bubbles in water is studied including interference and multiple scattering to all orders. These effects are studied through separate factors, one of which takes into account interference in the single scattering approximation, and the other accounts for multiple scattering effects. This allows the calculation of the total scattered field from the single scattering, incoherent field, which is easily calculated. The dependence of the effect on the array parameters and frequency is discussed. [Work supported by the Organization of the American States and Secretaría de Estado de Ciencias y Tecnología de Argentina.]

3:20

**L6. Ocean and model measurements of underwater acoustic scattering functions.** F. B. Tuteur, A. Zapinsky, and J. G. Zornig (Department of Engineering and Applied Sciences, Yale University, New Haven, CT 06520)

A series of measurements of the joint Doppler shift and delay (the scattering function) of surface scatter have been made under controlled conditions. Two series of experiments were conducted, one in the Pacific ocean and one in the model tank facility at Yale. The measurement technique in both cases was the use of a time series of probing pulses [J. G. Zornig, *J. Acoust. Soc. Am.* **64**, 1492-1499 (1978)] to extract fading time series data over a wide signal bandwidth. The result of an analysis of these data indicates a statistical relationship between frequency spread and relative arrival time which is not in agreement with previously reported theoretical results [F. B. Tuteur and J. F. McDonald, *J. Acoust. Soc. Am.* **57**, 1025-1029 (1978)]. The ocean and tank data are in substantial agreement and examples of a test of the data acquisition and analysis system, using solid moving targets, are presented as confirmation of method. [Work supported by NSF and MPL, Scripps.]

3:35

**L7. Bottom reverberation in the oceans.** Kenneth V. Mackenzie (Marine Science Consultants, P.O. Box 80715, San Diego, CA 92138)

Bottom reverberation generally follows Lambert's law for grazing angles less than 50° [P. B. Schmidt, *J. Acoust. Soc. Am.* **50**, 326-331 (1971)] with 10 log  $\mu$  between -24 to -34 dB with an average

near -28 dB obtained for 530- and 1030-Hz data [K. V. Mackenzie, *J. Acoust. Soc. Am.* **33**, 1498-1504 (1961)]. For steeper angles, scattering varies as  $\exp \cot^2 \phi$ , where  $\phi$  is the grazing angles [Y. Y. Jitkovskiy and L. A. Volovova, *E67*, 5th Int. Acoust. Congr. (1965)]. Reflection loss at normal incidence follows Rayleigh. A practical equation for a deep-ocean bottom reverberation model is presented and compared with available data.

3:50

**L8. Hydrographic reconnaissance of large undersea topography using scattered acoustic energy.** D. Schifter, E. Franchi, J. Griffin, and B. Adams (Naval Research Laboratory, Washington, DC 20375)

The TOPO test series was designed to measure reverberation from a variety of large-scale topographic features. From this research, an acoustic experimental and processing technique is developing which is applicable to wide-area hydrographic reconnaissance. Results from TOPO ONE are presented to illustrate the potential technique. A towed array receiver and explosive sources were used to obtain data covering a 1 000 000-square nautical mile area of a Pacific basin northeast of New Zealand. After filtering and beamforming, the processing provides spatial stabilization of beams and averaging over multiple shots to correct for left/right ambiguities and to create basin-wide mappings of the major scatterers. The resulting latitude-longitude display of the location and strength of major scatterers reveals both known and uncharted features. From TOPO ONE, isolated, uncharted seamounts appear within the basin and a section of the Louisville Ridge, previously contoured as an elongated plateau, is revealed to be a series of separate peaks. Subsequently, the New Zealand Navy has verified, by depth sounding, the existence of these features first identified in the acoustic data.

4:05

**L9. Influence of atmospheric pressure gradient on under-ice ambient noise.** Charles R. Greene and Beaumont M. Buck (Polar Research Laboratory, 123 Santa Barbara Street, Santa Barbara, CA 93101)

Correlations exceeding 50% have been measured between wind speed and under-ice ambient noise levels in one-third octave bands. Using barometric data, meteorologists derive geostrophic wind from the atmospheric pressure gradient. The present research involved measuring correlations between the magnitude of the atmospheric pressure gradient and under-ice noise levels in one-third octave bands to determine if noise-prediction models could be developed. Then, one could use meteorological data to generate "isonoise" contour maps for the Arctic Ocean similar to isobaric maps. Barometric and ambient noise data were available from a trio of buoys drifting in the Beaufort Sea for over a year. For one-third octave bands centered at 3.2, 10, and 32 Hz, correlations with the air pressure gradient magnitude were normally 50% or better. At 1000 Hz the correlation was nominally 25%, a result attributed to higher propagation loss at that frequency and the need for knowledge of pressure gradient on a smaller scale. Predictive models were derived for each frequency for each of the four annual sessions. [Work supported by ONR.]

4:20

**L10. Source level measurements of an arctic sea ice pressure ridge.** Beaumont M. Buck and Charles R. Greene (Polar Research Laboratory, 123 Santa Barbara Street, Santa Barbara, CA 93101)

Ice pressure ridge activity occurred over a three-day period in the immediate vicinity of a research ice camp 180 miles from the pole in the Eurasian Basin of the Arctic Ocean in 1979. The ridge was instrumented with both horizontal and vertical calibrated hydrophone pairs at a distance of about 100 m and the acoustic output tape recorded. Dimensional surveys were made atop the ice of the active portions of the ridge. Correlation and spectral analyses were per-

formed which allowed quantitative assessments of source level in the band 3–1000-Hz. These data, along with previously measured arctic transmission loss, were used to derive a model of pressure ridge area distribution and density by matching previously gathered

statistical ambient noise data. The observed ice ridge noise was abundant in fairly long-lasting tonals as low as 8 Hz attributed to a “slipping clutch” effect. These were the first known quantitative acoustical measurements of a nearby pressure ridge.

WEDNESDAY MORNING, 28 NOVEMBER 1979 • BONNEVILLE ROOM 2, 8:30 TO 10:45 A.M.

**Session M. Architectural Acoustics I: Reinforcement in Reverberant Spaces**  
(Session dedicated to Paul Boner)

David Klepper, Chairman

*KMK Associates Ltd., 96 Haarlem Avenue, White Plains, New York 10603*

✱  
Chairman's Introduction—8:30

*Invited Papers*

8:35

**M1. Paul Boner's contribution to the design, equalization, and operation of sound reinforcement.** Robert C. Coffeen (Coffeen, Anderson, and Associates Inc., 5805 Outlook, Mission, KS 66202)

The work of Dr. C. Paul Boner, relating to sound reinforcement systems for large spaces, ranks at the top of contributions in this field during the past two decades. Those who design and operate sound reinforcement systems, as well as those who simply listen to such systems, are indebted to Paul for his far-reaching efforts.

8:55

**M2. Gothic sound for the neo-gothic chapel of Duke University.** Robert B. Newman (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138) and James G. Ferguson, Jr. (Box 869, Chapel Hill, NC 27514)

The Duke University Chapel, a 250-ft-long, one million cu ft neo-gothic structure, was built in 1930–32. It was finished on half of its interior surface with Akoustolith, a precast artificial sound absorbing stone. The remaining surfaces were Indiana limestone. This resulted in a mid-frequency reverberation time of about 3 s in the unoccupied space. A donor offered to give a new organ for the chapel provided the space could be made to sound the way it looks. After considerable experimentation, four coats of sealer were applied to the Akoustolith. The reverberation time was increased to over 7 s, creating an ideal environment for the new organ. The old speech reinforcement system was then hopeless as had been predicted. A new system was designed and installed. It uses column loudspeakers at each structural pier in the nave with additional column speakers serving crossing and transepts, all with appropriate time delays. Now the chapel has superb conditions for liturgical music and excellent speech intelligibility.

9:15

**M3. The University of Texas at Austin special events center: acoustics, noise control, and audio systems.** Charles R. Boner and Kenneth R. Dickensheets (Boner Associates, Consultants in Acoustics, Austin, TX 78763)

This paper explores the acoustic design, noise control design, and sound/audio system philosophy behind this 18 000-seat, multi-purpose arena. The paper will focus in particular on the unique manner in which the interaction between the Consultant and Building Manager helped create an ideal environment for a state-of-the-art building. A complete description of the acoustical treatments, noise control design, and audio systems and equipment will be presented.

9:35

**M4. Acoustical environment in the Salt Lake Mormon Tabernacle.** L. Dean Jones, Karl G. Seljaas, and David E. Drommond (The Church of Jesus Christ of Latter-day Saints, 50 East North Temple Street, Salt Lake City, UT 84150)

This paper is a historical overview of the Salt Lake Mormon Tabernacle, its acoustics and uses. Emphasis will be placed on acoustical studies made over the years by qualified consultants and their subsequent recommendations. Reference will be made to a previous paper on this subject presented nearly 50 years ago. Slides will be used to illustrate original construction and later changes. Television lighting and sound reinforcement equipment are substantially the only detractors from an otherwise unspoiled atmosphere of pioneer days.

9:55

**M5. Loudspeaker array design for a high school gymnasium.** J. Robert Ashley and John Charles Cox (University of Colorado at Denver, 1100 Fourteenth Street, Denver, CO 80202)

Boulder High School gymnasium, built in 1937 for basketball, has used two P.A. systems of unsatisfactory performance. The usual budgetary situation requires that an attempt be made to install equipment which does not include deadening of several badly reflecting surfaces. To prevent loss of intelligibility because of auditory backward inhibition, the loudspeaker array must illuminate the seating area with minimum illumination of the reflecting surfaces. An estimate of the cost of an adequate horn array quickly ruled out this approach. The work on electrically tapered linear array loudspeaker systems by the Australian, Dr. J. E. Bensen, was adapted to the design of a loudspeaker array which is electrically tapered in both horizontal and vertical directions. Measured directivity of this array will be presented. Performance of the installation will be discussed.

### Contributed Papers

10:15

**M6. Sound reinforcement for the Pontiac Silverdome.** Robert C. Coffeen (Coffeen, Anderson, and Associates Inc., 5805 Outlook, Mission, KS 66202)

After several years of use, the sound reinforcement system for the Pontiac Silverdome has demonstrated its capabilities for football, basketball, and other uses. The Pontiac Silverdome has a volume of approximately 49 million cu ft, and the unoccupied mid-frequency reverberation period is approximately 9 s. The conditions provide a difficult environment in regard to the provision of suitable sound reinforcement.

10:30

**M7. Speech reinforcement in a large cathedral, at the limits of D.** L. K. Irvine and R. K. Fullmer (Acoustical Engineers Inc., 1864 South State, Salt Lake City, UT 84115)

A state-of-the-art point source speaker system was installed in a large (627 000 cu ft) cathedral, with 3.5 s  $RT_{60}$ . Initial calculations using D. Davis' theory for articulation are compared with results of nonsense word articulation tests made in the building using the installed system. Also, results of octave band  $RT_{60}$  measurements are plotted, showing results of pioneering acoustical corrections performed in 1916 using covered felt blankets applied to ceiling.

WEDNESDAY MORNING, 28 NOVEMBER 1979 BONNEVILLE ROOM 4, 8:30 TO 11:05 A.M.

Session N. Physical Acoustics III, Engineering Acoustics III, and Noise IV: Acoustic Remote Sensing II

C. I. Chessell, Chairman

*Cooperative Institute for Research in Environmental Sciences, University of Colorado, Boulder, Colorado 80309*

Chairman's Introduction—8:30

### Contributed Papers

8:35

**N1. Rotary spectrum analysis of sodar Doppler signals.** K. H. Underwood (Department of Meteorology, Pennsylvania State University, University Park, PA 16802)

Rotary spectrum analysis is a proven method for interpretation of geophysical data exhibiting inherent rotational characteristics. Computationally derived using a discrete Fourier transform, it shows the extent to which contributions to the total variance originate from

clockwise or counterclockwise rotating components. The technique is well suited to the analysis of Doppler sodar signals which are sampled as the in (I) and quadrature (Q) phase components of a rotating phaser. The Penn State Doppler sodar system may be operated either in a "track" or I and Q mode. This paper demonstrates the advantages of I and Q sampling and rotary spectrum analysis for detailed interpretation of atmospheric wind and turbulence profiles. A comparison of Doppler-derived and 123-m tower measured winds will be presented.



8:50

**N2. Computerized sodar for network measurements of planetary boundary layer structure.** R. L. Peters and D. W. Thomson (Department of Meteorology, Pennsylvania State University, University Park, PA 16802)

Local or regional scale analysis of planetary boundary structure, such as is required for gravity wave or mixing depth analyses, can be greatly facilitated by using a network of sodar systems. Such a network will typically include three to twenty acoustic sounders and some supplementary sensors such as sensitive electronic microbarographs. Use of network observations is greatly simplified if the distributed sodars are controlled by the measurements logged on the display at a central minicomputer. This paper presents an inexpensive, telephone network system designed for sodar control and data acquisition. The combination digital and analog signal transmission was optimized for telephone transmission of sodar control and signal data. Features include automatic answering and hangup, simultaneous sodar triggering, data acquisition from all network sites, and automatic remote calibration of the interface system. The system requires only the availability of a standard-voice Bell system compatible circuit.

9:05

**N3. Atmospheric structure and migrating birds: Acoustic echo sounder and tracking radar investigations.** Ronald P. Larkin (The Rockefeller University, New York, NY 10021)

The migrations of birds, often taking place at night and spanning hundreds or thousands of kilometers, involve intimate contact with the lower atmosphere. Structure in the atmosphere is hypothesized to be a source of information in avian long-distance navigation and also an important source of extractable energy. Studies of the use of atmospheric structure by migrating birds employ a monostatic atmospheric echo sounder and a 3-cm radar, the latter allowing quantitative description of spatial concentrations of migrating birds and also tracing wavelike phenomena using balloon-borne radar targets. Results include: (1) The environmental impact of a pulsed acoustic sounder upon nocturnally migrating birds is minimal except for birds directly within the beam. (2) Altitudes of migrating birds are sometimes closely related to stable regions of shear appearing on the acoustic sounder records. (3) Vertical motions exceeding 1 m/s are frequent on nights when birds are migrating, providing great potential benefit to birds (many flying slower than 5 m/s horizontal speed) that detect and selectively fly in regions of rising air. [Work supported by NSF.]

9:20

**N4. Regional air pollution study acoustic radar analysis program.** T. L. Waldron (Rockwell International Environmental Monitoring and Services Center, 11640 Administration Drive, Creve Coeur, MO 63141) and J. L. McElroy (U.S. EPA, Box 15027, Las Vegas, NV 89114)

An acoustic radar (sodar) was operated in urban St. Louis from February 1975 through May 1977 as part of the Regional Air Pollution Study (RAPS). Comprehensive analysis procedures were developed to catalogue the acoustic data and supporting meteorological data in a generalized format but with sufficient detail to be utilized for various RAPS studies. The described analysis program includes assignment of one of thirty documented echo pattern classifications, determination of base and top heights for up to three layers each half hour, and echo description codes for intensity, height tendency, and boundary appearance. Meteorological data input includes 24-h narrative summaries, identification of frontal passages, 3-h surface data, and 12-h 850-mb data. A typical "plain language" output is presented. Slow ascent radiosondes were released at the study site every 6 h during the entire operational period allowing extensive comparison between the vertical profile data and the

acoustic echo pattern types. Results are presented for one stable and one unstable classification.

9:35

**N5. Acoustic propagation in a thermally inhomogeneous atmosphere.** W. K. Van Moorthem and James R. Butterworth (Mechanical and Industrial Engineering Department, University of Utah, Salt Lake City, UT 84112)

A series of propagation experiments have been carried out at distances between ¼ and 1 mile on a dry lake bed in western Utah. Detailed measurements were made of the vertical temperature profile within about 6 m of the ground along with some low-level wind-speed measurements. Although propagation both upwind and downwind was considered, the strong temperature gradients near the ground would have resulted in the formation of an acoustic shadow in all cases considered. The results obtained, however, show a clear correlation with wind direction, similar to the upwind and downwind cases of earlier investigators. It appears that winds above the level measured here play a significant role in long range propagation. [Work supported by NSF.]

9:50

**N6. Determination of excess ground attenuation for shallow angles using flyover noise.** William L. Willshire, Jr., and David A. Hilton (National Aeronautics and Space Administration, Langley Research Center, Hampton, VA 23665)

An outdoor experiment was conducted during the week of 30 October–4 November 1978 at NASA Wallops Flight Center to investigate the propagation of sound near grazing incidence. A T-38A aircraft was flown at low altitudes over the ends of two microphone arrays. An eight-microphone array was positioned along a 1850-m concrete runway. The second array consisted of 12 microphones positioned parallel to the runway over grass. Twenty-eight flights were flown at altitudes ranging from 10 to 160 m. The corresponding angle-of-incidence range for the microphone arrays used was 0.3° to 32°. The aircraft was tracked with a laser tracker and meteorological information was collected from six ground stations and a retrievable balloon instrumented for weather parameter profile measurements. The acoustic information recorded in the field was reduced by ½-octave-band techniques and time correlated with the flight and weather information. The data have been further reduced to values of excess ground attenuation as a function of frequency and look angle by two different methods. In one method, referred to as the method of Parkin and Scholes, acoustic data at a microphone position close to the flight track is chosen as a reference and is compared with those data taken at other microphone positions. A direct comparison between two microphones at an equal distance from the flight track, but over different surfaces, was the second method used. In both methods the source emission directivity angle was the criterion used to select the portions of the acoustic signals to be compared. The paper will conclude with a comparison of the measured values of excess ground attenuation from each method.

10:05

**N7. Effects of ground cover on the propagation of sound through the atmosphere.** L. N. Bolen and H. E. Bass (Department of Physics, University of Mississippi, University, MS 38677)

Measurements of sound amplitude in the vicinity of a ground plane have been made as a function of frequency of the sound source (50–2000 Hz), distance of propagation (5–300 m), and surface conditions. By treating the impedance as an adjustable parameter, the surface impedance as a function of frequency was determined from the measured amplitudes using a theoretical treatment of a spherical wave in the vicinity of a locally reacting surface. The

impedance measurements covered the frequency range 50–1000 Hz. In this frequency range, the results for three distinctly different surfaces suggest that the impedance can be computed from the specific flow resistance and that grass has little effect on the surface impedance except for decreasing the flow resistance due to the root structure. The results of the values of impedance calculated from sound measurements and flow resistance measurements will be compared, and the applicability of the single parameter model of Chessell to the data will be discussed. [Work supported by U.S. Army Research Office.]

10:20

**N8. Acoustic propagation over ground; a computer study.** G. K. Miller and F. E. Babian (GTE Sylvania, Box 188, Mountain View, CA 94042)

A computer model was developed for the acoustic propagation between a sinusoidal source and a receiver when both are located on or above the planar surface of a homogeneous medium of known specific flow resistance. The "locally reacting" assumption is made and the "exact" formulation of Thomasson is numerically integrated. The normal specific acoustic impedance of the ground is obtained using the empirical model of Delany and Bazley. Atmospheric absorption is included as a function of local temperature, relative humidity and barometric pressure, and the effects of the source–receiver geometry, (source and receiver heights and their horizontal separation) are variable inputs. Provisions for approximating non-point sources and for accounting for the presence of wind and temperature fluctuations are being made. Current results clearly show many examples of the importance of the ground effect, and the source–receiver geometry, on the excess attenuation as a function of frequency.

10:35

**N9. Farfield distortion of transient waveforms propagating through the atmosphere.** Allan D. Pierce (U.S. Department of Transportation, Transportation Systems Center, Cambridge, MA 02142)

It has long been known [E. F. Cox, in *Encyclopedia of Physics*, (Springer–Verlag, Berlin, 1957), Vol. 48, pp. 469, 472] that transient infrasound pulses, received at horizontal distances of 20 km or larger from their source, exhibit substantial changes in signature (e.g., more cycles) from those received closer in. Nonlinear effects, relaxation effects, and ground impedance effects are shown to be inadequate to account for this distortion. The author's explanation rests on the presence of atmospheric turbulence and is similar to the wavefront folding mechanism [J. Acoust. Soc. Am. 49, 906–924 (1969)] previously invoked to explain the anomalous rise times of sonic booms. The turbulent winds cause any single geometrical acoustics ray predicted for a perfectly stratified atmosphere to be replaced by a family of closely spaced rays connecting source and listener. Diffraction washes out turbulent scales less than a representative wavelength divided by  $2\pi$ . The further the listener, the more such rays and the more cycles in the waveform. It is suggested that an analysis of the distorted waveform will yield a parameter (analogous to Crow's  $t_c$ ) that gives a gross integrated average measure of the turbulence along the nominal ray path.

10:50

**N10. Model studies of aircraft sideline noise.** C. I. Chessell (Cooperative Institute for Research in Environmental Sciences, University of Colorado, Boulder, CO 80309)

The prediction of aircraft sideline noise from overhead data requires a quantitative understanding of the excess attenuation introduced by the presence of the ground between the aircraft and observer. This attenuation becomes particularly important when the aircraft subtends low elevation angles relative to the observer. A number of gross empirical models have been developed for this phenomenon based on limited experimental data. In this paper theoretical predictions of excess ground attenuation are presented. The fibrous absorbent material model is used to represent the finite acoustic impedance of the surface. Excess ground attenuation as a function of range, elevation angle, surface impedance, and aircraft source spectrum is studied and the theoretical results compared with experimental measurements of sideline noise.

WEDNESDAY MORNING, 28 NOVEMBER 1979 BONNEVILLE ROOM 1, 9:00 TO 10:53 A.M.

## Session O. Musical Acoustics II: General Musical Acoustics

Thomas D. Rossing, Chairman

*Northern Illinois University, Dekalb, Illinois 60115*

Chairman's Introduction — 9:00

### *Contributed Papers*

9:05

**O1. Radiation fields of a violin in the region of the Helmholtz and main body resonances.** Gabriel Weinreich and Eric B. Arnold (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

The measuring system previously described [G. Weinreich and E. B. Arnold, J. Acoust. Soc. Am. Suppl. 1 65, S72(A) (1979)] has been used to analyze the acoustic fields radiated by a violin in the frequency range of the Helmholtz and main body resonances. The violin is driven electro-dynamically by an ac current through

the string and a permanent magnet mounted over it. The frequency of the driving oscillator is locked to the string resonance by a feedback loop actuated by an acceleration sensor at the bass foot of the bridge, and is adjusted by moving a small Delrin mass which slides on the string, so that the tension maintains its normal value. Observations at the Helmholtz resonance include the near-field motion of air through the  $f$  holes, which is otherwise difficult to see, and the resulting farfield pattern. Because of the  $90^\circ$  relative phase shift between the  $s$  wave and  $p$  wave as one moves from the nearfield to the farfield, the nearfield pattern, with its strong motion in the forward direction, becomes completely bidirectional in the farfield. [Work supported by NSF.]

**O2. Calibration procedure and error analysis for a violin radiation measuring system.** Eric B. Arnold and Gabriel Weinreich (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

The system referred to in the preceding paper involves measurement of acoustic pressure, both in amplitude and in phase, on two concentric spheres of known radii. Among possible sources of error are: the finite number of measurements; inaccuracy in microphone positioning; scattering by the moving boom system; electrical and acoustical noise in the microphone system; spurious phase shifts and gain inequalities in the microphone and amplifier channels; timing errors and overtone sensitivity of the phase-sensitive detectors; offsets and quantizing errors in the analog-to-digital converter; and numerical computation errors. Effects due to the nonideal nature of the anechoic chamber are in a different category, since our system measures the reflected wave directly. [Work supported by NSF.]

9:29

**O3. Measured reproducibility of clarinet spectra.** A. H. Benade (Case Western Reserve University, Cleveland, OH 44106) and C. O. Larson (University of Wisconsin, River Falls, WI 54022)

Published spectra show little consistency beyond a tendency toward strong odd partials. Neglect of radiation pattern and wave-statistics fluctuations in rooms, join with inadequate specification of the player's task to cause this variability. We have studied 0.5 s alternating  $C_4$ ,  $D_4$  tones lasting about 35-s total at mezzoforte level (reed just beginning to beat), played on six clarinets, A,  $B\flat$  (three), C,  $E\flat$ . All instruments were optimized to similar criteria over the past decade by one of us (AHB) who also made each its own mouthpiece (the bores were dissimilar). The instruments all have been used by major players. The player (AHB) walked about while playing (0.75 m/s), as did the two persons carrying the recording microphones, in a room  $V = 8000 \text{ m}^3$ ,  $T_{60} = 1.2 \text{ s}$ . Source and mike position changes, plus mode perturbation effects from three people, assure the equivalent of at least 75 statistically independent data points for each component of each tone. The SPL for the first five harmonics averaged over all twelve tones are 50, 22.6, 48.8, 27.0, 48.3 dB ( $\sigma = 2 \text{ dB}$ ). Room fluctuations are within  $\pm 1 \text{ dB}$  (95% confidence). Most of the remaining variation comes from interinstrument differences and not player/instrument instabilities. These results are consistent with a preliminary experiment analyzed by Karla Brasaemle. [Work assisted by NSF.]

9:41

**O4. A functional model of a simplified clarinet.** S. E. Stewart (Code 5311, Naval Ocean Systems Center, San Diego, CA 92152) and W. J. Strong (Brigham Young University, Provo, UT 84602)

A functional model (computer simulation) of a simplified clarinet has been implemented on a digital computer. The simplified clarinet consists of a standard clarinet mouthpiece and reed attached to a straight cylindrical tube without tone holes. The tube and mouthpiece are represented in the model by a lumped element approximation to a transmission line. Application of Kirchhoff's circuit laws to this circuit gives a coupled set of differential equations which were solved numerically on a digital computer to yield volume velocities and pressures in the mouthpiece and tube. The pressure in the mouthpiece drives the reed which is represented in the model as a tapered bar clamped at one end. The partial differential equation describing vibrations of such a bar was solved numerically to obtain the size and shape of the reed aperture. The reed aperture area governs the volume velocity into the tube which in turn drives the air column vibrations. The model exhibits self-sustained oscillations, threshold blowing pressures, and "lipping" of the tone. The spectra of the mouthpiece- and radiated-pressures are in general

agreement with actual clarinet spectra. Some interesting features of the model are the inclusion of a frequency-dependent viscous loss in the tube and a dependence of the aperture volume velocity on the initial or rest opening of the aperture.

9:53

**O5. Temperature-induced length and tension variations affecting the pitch of a stretched string.** Robert E. Kelly (Physics Department, University of Mississippi, University, MS 38677)

The pitch variation of a stretched string is analyzed in terms of length and tension perturbations caused by temperature. In particular, it is shown that tension changes due to differential expansion between the string and its frame usually dominate the effects caused by an explicit length deviation. Numerical examples are given for various combinations of string and frame compositions. The corresponding problem for the wind instruments is briefly considered, including the typical viewpoint of performing musicians.

10:05

**O6. Contrasting sounds in the upper male voice.** Lloyd A. Smith (Department of Speech Communication and Drama, North Texas State University, Denton, TX 76203)

Four sounds were contrasted in the upper male voice. The sounds were identified as follows: (1) falsetto; (2) head tone; (3) pharyngeal voice; (4) operatic head voice. All sounds were sung by a tenor on the vowel /a/ at a pitch level of  $A_4$ , 415 Hz. Spectrographic analyses were performed to determine what physiological relationships might be inferred among the sounds. The length of time the singer was able to sustain each sound was tested in order to gain information concerning the relative air flows. The results show that the pharyngeal voice and operatic voice have much more energy in the high partials and have much lower rate of air flow. This evidence indicates that these two sounds are characterized by much more complete closure of the glottis than is exhibited in the other two sounds. It is concluded that the four sounds are produced by only two different laryngeal adjustments. Acoustical differences accounting for differentiation of sounds produced by the same laryngeal setting are discussed. A recommendation is made concerning the pedagogical use of registers and suggestions are made for further research.

10:17

**O7. The effect of envelope on fusion of tones with inharmonic partials.** Elizabeth Cohen (Graduate Special Studies: Acoustics, Room R123, Stanford Medical Center, Stanford, CA 94305)

Temporal envelopes for tones with an inharmonic partial structure were generated using the Systems Concepts Digital synthesizer. [E. Cohen, J. Acoust. Soc. Am. Suppl. 1, 65, S123(A) (1979)]. The reasons for choosing envelopes based on musical viability rather than a flat "psychoacoustically standard" pattern, will be discussed. Subjects were asked to make judgments on the degree of tone dispersion for single tones differing in degree of inharmonicity and temporal envelope. The importance of attack and decay times, as well as transients in the signal will be explained. A real-time experiment on the determination of subjective interval size of dyads using two different envelope patterns also revealed that duration of the "steady state" portion of signal has an effect on tone fusion.

10:29

**O8. Optional modules for a flexible musical acoustics course.** Donald E. Hall (Physics Department, California State University, 6000 J Street, Sacramento, CA 95819)

Musical acoustics courses draw students with a wide range of backgrounds, from nonmusicians to graduate music majors. Within either group there is a variety of topical interests—one instrument or another, room acoustics, hi-fi, harmonic theory, perception, etc. To meet the needs of such a diverse group, I have been offering a course with a spiral organization that branches into roughly a dozen optional modules, of which each student is to choose two or three for detailed study. I will describe the course outline and how the module assignments are handled. Student evaluations of this approach are very favorable, and have also given useful feedback for improving the scheme.

10:41

**09. Loudspeaker requirements for electronic organs in churches.** J. Robert Ashley and Roy A. Pritts (University of Colorado at Denver, 1100 Fourteenth Street, Denver, CO 80202)

When visiting a strange church, we can identify the organ as a pipe organ or an electronic organ with just a few minutes of listening (and without visual inspection). A run on the pedal organ will show the usual organ loudspeaker to cease radiating below 60 Hz. Modulation distortion will show that the all important middle octave (261–525 Hz) is being radiated from the same cone as the low tones. The lack of high-frequency power radiation is noticeable. We deem these faults are caused by the fact that organ loudspeakers have not been designed, they have just happened. Dr. J. E. Bensen reported (in 1959) an experiment in the Sydney (Australia) Town Hall which proves this need not be true. We have repeated Dr. Bensen's experiment in the First Methodist Church, Colorado Springs, Colorado, and verified his conclusion. To prove that the recent loudspeaker revolution can make the electronic organ viable, we use a Moog Synthesizer to simulate several pedal and great organ stops.

WEDNESDAY MORNING, 28 NOVEMBER 1979 BONNEVILLE ROOM 5, 9:00 TO 11:05 A.M.

### Session P. Noise III: Community Noise

Angelo J. Campanella, Chairman

*Acculab, 3201 Ridgewood Drive, Columbus, Ohio 43220*

Chairman's Introduction—9:00

#### Contributed Papers

9:05

**P1. Statistical independence of a series of ambient noise measurements.** D. A. Driscoll, (NYS Department of Public Service, The Governor Nelson A. Rockefeller Empire State Plaza, Albany, NY 12223) and W. J. Webster (NYS Department of Environmental Conservation, 50 Wolf Road, Albany, NY 12233)

The autocorrelation of a series of noise parameters ( $L_{eq}$ ,  $L_{10}$ ,  $L_{50}$ , or  $L_{90}$ ) was examined. The noise parameters were derived from 15-min records of ambient noise collected at the same hour (0000, 0600, or 1800 h) on each of 100 (predominantly consecutive) days at a rural site. The same analysis was also performed for a suburban and a quiet, urban, residential site. The first zero crossing (indicating statistical independence) occurs at 9–13 days for the rural site. A run test for randomness (0.05 level of significance) was also used. The random hypothesis consistently failed for the rural site, showing that consecutive days do not produce a random string of data. It is shown that for an efficient ambient survey, samples should be separated by at least three days for rural sites, but that consecutive day samples are normally satisfactory for urban and suburban sites.

9:20

**P2. Compatibility in federal community noise policies.** Mones E. Hawley and Richard J. Peppin (Jack Faucett Associates, 5454 Wisconsin Avenue, Chevy Chase, MD 20015)

Several federal agencies have community noise policies. These agencies include the Environmental Protection Agency, the Federal Aviation Administration, the Federal Highway Administration, the Department of Defense, the Department of Housing and Urban Development, and the General Services Administration. In some cases, the policies are consistent, compatible, and complete. In some cases, they are inconsistent and, in other cases, the coverage is incomplete.

This paper describes and compares these policies, their implications for enforcing agencies, and their effects upon citizens.

9:35

**P3. Community noise ordinance drafting by a city ad-hoc noise committee.** Angelo J. Campanella (Acculab, 3201 Ridgewood Drive, Columbus, OH 43220) and P. Gary Brown (Columbus Department of Health, Columbus, OH 43215)

Through motivation by a city councilman, Charles Petree, an ad-hoc committee was formed to review existing city noise-related codes. Representation included city council, the health, zoning, and police departments, an acoustical consultant, the industrial commission and building association, the society of professional engineers, the electric company, and a private citizen. This body met many times over one year to review codes and ordinances. Recommendations included consolidating all noise related codes and ordinances into one addendum to the City Code. Additions against squealing tires, limits on motorboat noise, and the replacement of octave-band limits for manufacturing districts with  $L_{eq}$  limits along common residential/institutional/commercial/manufacturing property lines were made. The police department concluded that the existing muffler ordinance requiring "a muffler in good working condition with baffle plates . . ." was satisfactory for enforcement. These recommendations—to be detailed in the presentation—are being submitted to the City Council for action. Airport noise, as it affects land use and zoning actions, will be treated during calendar year 1980.

9:50

**P4. An airport community noise-impact assessment model.** Richard DeLoach (Noise Effects Branch, NASA Langley Research Center, Hampton, VA 23665)

A computer model has been developed to assess the noise impact of an airport on the community which it serves. Assessments are made using the Fractional Impact Method by which a single number describes the community aircraft noise environment in terms of exposed population and multiple-event noise level. The model is comprised of three elements: a conventional noise footprint model, a site-specific population distribution model, and a dose-response transfer function. The footprint model provides the noise distribution for a given aircraft operating scenario. This information is combined with a site-specific population distribution obtained from a national census data base to yield the number of residents exposed to a given level of noise. The dose-response relationship relates noise exposure levels to the percentage of individuals who would describe themselves as "highly annoyed" by those levels. This information is used to compute a single-number descriptor of the airport noise environment. In addition to providing a quantitative assessment of the noise environment in the community at large, the model generates a report which lists several demographic variables as a function of noise level which are of interest to community planners and others. These variables include population density, growth rate, average age, average home value, percent homeowners, percent renters, and others. This paper describes the structure and operation of the community response model and presents the results of initial noise impact assessment studies.

10:05

**P5. Cumulative annoyance due to multiple aircraft flyovers with differing peak noise levels.** K. P. Shepherd (Bionetics Corp., 18 Research Drive, Hampton, VA 23666)

A laboratory experiment was conducted in which 160 subjects made annoyance judgments of half-hour sessions of aircraft flyover sounds. Each of ten test sessions contained nine flyovers with various peak noise levels such that the session  $L_{eq}$  values had a range of 20 dB. After each session, the subjects assessed their annoyance in the laboratory and also estimated their annoyance to such a noise environment occurring in their homes during the day, evening, and night. The laboratory annoyance judgments were examined in terms of their relationship with various proposed noise rating scales, from which it was concluded the energy summation scales such as  $L_{eq}$  and Noise Exposure Forecast were superior to the peak dB(A) concept [R. Rylander, *J. Sound Vib.* **36**(3), 399-406 (1974)] and to Noise Pollution Level [D. W. Robinson, *Nat. Phys. Lab. Aero Rep. Ac38* (1969)]. The subjects' responses to the projected annoyance questions (time of day weighting) were compared with aircraft noise social survey results. [Work supported by NASA Langley Research Center.]

10:20

**P6. Effect of unit load and fan speed on induced-draft fan noise at electric generation stations.** A. R. Thompson (Stone & Webster Engineering Corp., Boston, MA 02107) and J. P. Buechler (Long Island Lighting Company, Hicksville, NY 11801)

Induced-draft fan noise has been recognized as a major source of community noise in the electric power industry. Typically, the dominant path of ID fan noise is from the top of the stack with the noise being both tonal and broadband when centrifugal fans are used. Depending upon site-specific factors, the noise can be a problem at full-load operating conditions. At low-load conditions, the problem can become more severe as airflow is reduced by closing the inlet dampers. This results in higher levels of the tones and the broadband noise. A series of measurements, taken at three units at various load conditions, indicate that the levels of the tones, particularly the second harmonic, increase as the load is decreased. At low loads, when the dampers are relatively closed, significant reductions in the tone levels can be achieved by switching to a lower fan speed. A lower fan speed results in reduction of the tip speed and opening of the dampers for a given load condition.

10:35

**P7. Construction noise control for a major urban rapid transit line extension.** W. J. Cavanaugh and G. C. Tocci (Cavanaugh Tocci Associates, Natick, MA 01760) and T. A. Muldoon (Massachusetts Bay Transportation Authority, Boston, MA 02140)

The northwest extension of the Massachusetts Bay Transportation Authority Red Line Rapid Transit System involves the demolition and reconstruction of the current terminus station—Harvard Square Station, Cambridge—and the construction of three new stations, and associated subway tunnel and track work. This major construction program, scheduled to extend over a five year period, recognizes the potential noise impacts on the densely populated and highly developed communities in which the new construction is to take place. Cambridge, the most significantly affected community, currently has a noise ordinance limiting property line  $L_{10}$  and maximum A-weighted sound levels due to construction operations. The MBTA has incorporated the Cambridge limits into construction contract documents and has also included limits on noise produced by individual pieces of construction equipment. The construction noise specifications and implementation procedures are described along with the experience to date in actual contract applications. [This work is partially supported by the US Urban Mass Transit Administration and the Massachusetts Bay Transportation Authority.]

10:50

**P8. The effect of stoplights on traffic noise.** R. E. Halliwell (Division of Building Research, National Research Council of Canada, Ottawa, Canada K1A 0R6)

A study was undertaken to determine how the noise level due to free-flowing traffic is affected by the insertion of a traffic light. Field measurements were taken at eight different traffic light locations, representing two configurations; that of two intersecting straight roads and tee junctions. A reference level, measured at a point where traffic noise was unaffected by the intersection, was used in conjunction with the NRC traffic noise prediction model to assess the change in noise level in the region about the traffic light.

**Session Q. Physiological Acoustics II: Central Auditory System**

Jerry L. Cranford, Chairman

*Department of Otolaryngology, University of Texas Medical Branch, Galveston, Texas 77550*

Chairman's Introduction—8:45

**Contributed Papers**

8:50

**Q1. The influence of own-phonation on parts of the central auditory pathway in the squirrel monkey.** P. Müller-Preuss (Max-Planck-Institut für Psychiatrie, München, Deutschland)

The reactivity of the auditory cortex and the auditory midbrain to self-produced and loudspeaker-transmitted calls were investigated in order to study the relationships between brain structures which are involved in the production of vocalizations and brain structures which take part in auditory processing. Responses of single cells were extracellularly recorded during stimulation with self-produced vocalizations (elicited mainly through electrical brain stimulation or uttered partially spontaneously), tape-recorded playback of those self-produced vocalizations and other species-specific vocalizations. Over 50% of the cortical cells which respond to loudspeaker transmitted calls, gave no response to self-produced vocalizations or a significantly weaker one. In contrast, most of the midbrain cells tested show no difference in their reactions to self-produced and loudspeaker transmitted calls. The response patterns of units which respond to self-produced cells do not differentiate between self-produced and playback vocalizations. It is concluded that brain structures which are activated through phonation can have an inhibitory influence on central auditory structures and that this influence takes place at a higher level than the auditory midbrain.

9:05

**Q2. Discrimination of brief tones by humans with temporal lobe damage.** J. L. Cranford and R. W. Stream (Department of Otolaryngology, University of Texas Medical Branch, Galveston, TX 77550)

A number of recent Russian investigations [A. V. Baru and T. A. Karaseva, *The Brain and Hearing*, Consultants Bureau, New York, (1972)] reported that dogs and humans with large auditory cortex lesions, exhibit significant increases in both their absolute detection thresholds as well as their frequency difference limens for tonal signals which are shorter than 16 ms in duration. In contrast, more recent studies [J. L. Cranford, *J. Acoust. Soc. Am.* **65**, 1573–1575 (1979)] with cats suggest that the cortex, rather than being essential for detecting the simple presence of brief tones, may be more important for discriminating qualitative differences in such sounds. While having normal absolute detection thresholds for brief tone pulses, cats with auditory cortex lesions, in comparison to unoperated controls, do exhibit elevated frequency difference limens. In order to investigate the inter-species generality of these new animal findings, we recently presented the same series of brief tone tests to two human patients with temporal lobe lesions (one left-sided CVA, and one right-sided glioblastoma case). The results with the two patients are remarkably similar to those obtained with cats. Both patients exhibited normal absolute detection thresholds for brief tones in combination with significantly elevated frequency difference limens. [Work supported by Deafness Research Foundation and NINCDS.]

9:20

**Q3. Spectral basis of human frequency-following response to the "missing fundamental".** Steven Greenberg (Department of Linguistics, University of California, Los Angeles, CA 90024) and James T. Marsh (Department of Psychiatry and Brain Research Institute, University of California, Los Angeles, CA 90024)

Psychophysical studies of the "dominance" and "existence" regions of the "residue" underscore the importance of peripheral frequency analysis in the perception of pitch. However, the means by which the peripheral analysis of individual frequency components is transformed into a unitary percept dominated by the pitch of the residue, remains obscure. Farfield, vertex-derived, frequency-following responses (FFR), representing the integrated synchronized activity of phase-sensitive neurons most likely situated in the inferior colliculus, were recorded during presentation of multicomponent ( $n = 2, 3, 4$ ), harmonically related tone bursts. The frequency spectrum and waveform periodicity of FFR to low-order harmonics (harmonic ranks 3–5) are clearly dominated by a frequency corresponding to the "missing fundamental". The amount of energy in this frequency band diminishes considerably with increasing harmonic rank, becoming negligible for harmonics above the thirteenth. The existence of a parallel between psychophysical and FFR measures of pitch strength raises the possibility that in the brainstem auditory pathway, the "transmission code" for residue pitch is based on neural periodicity.

9:35

**Q4. Abstract withdrawn.**

9:50

**Q5. Dorsal cochlear nucleus (DCN) and its contributions to the brain stem electric response (BSER).** Jack Markuszka and Allen F. Ryan (Section of Otolaryngology, VA Hospital, La Jolla, CA 92161 and Division of Otolaryngology, UCSD Medical School, San Diego, CA 92037)

This report focuses on the contributions of the DCN to the BSER as part of a larger study of the origins and clinical applications of the BSER. Since their initial description, the successive waves of the BSER were thought to be related to the successive nuclei of the brain stem auditory pathway. Although a strict correspondence appears not to be the case, the exact nature of the physiology and underlying anatomy of the BSER is very much in doubt. Restricted lesions of the cochlear nucleus were utilized to explore the role of various elements within the nucleus in the generation of the BSER, as well as to assess the sensitivity of the BSER to restricted lesions of the central auditory pathway. DCN was selected as the first independent variable because it is easily accessible for aspiration lesions. The results suggest that (1) an overall reduction of the BSER occurs as a result of general trauma; (2) waves I, II, and V are not otherwise affected by destruction of DCN; (3) waves III and IV are altered after

DCN lesions. The nature of the changes, their clinical implications, and future directions of our research will be discussed. [Supported by the Research Service of the Veterans Administration.]

10:05

**Q6. Acoustic stapedial reflex latency versus BERA wave V latency in the detection of eighth nerve lesions.** C. N. Sarno and J. D. Clemis (Otologic Neurophysiology Laboratory, Mercy Hospital, Chicago, IL 60616)

The parameter of latency of the acoustic stapedial reflex was used to test the hypothesis that a retrocochlear lesion involving the afferent portion of the reflex arc will result in a slowed neural conduction time. Measurements of middle-ear muscle activity were made by use of an electro-acoustic impedance bridge modified to permit presentation of tonal stimuli 300 ms in duration with 10-ms rise/fall time for test frequencies of 1000 and 2000 Hz. Both contralateral and ipsilateral values were obtained for absolute reflex latency and interaural latency differences on four clinical populations which included cochlear lesions due to Meniere's disease and surgically confirmed acoustic tumor. Results reveal a dramatic prolongation of latency in the presence of retrocochlear lesions. Comparisons drawn between the acoustic reflex latency test and BERA wave V latency, obtained with tone pip stimuli, indicate that stapedial reflex latency is a more efficient method of tumor detection. [Work supported by American Hearing Research Foundation, Chicago, IL.]

10:20

**Q7. Electrical stimulation in the descending auditory system.** J. F. Glenn and L. C. Oatman (U.S. Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005)

An investigation of the effects of electrical stimulation in the descending auditory pathways on the latency and amplitude characteristics of auditory evoked potentials was carried out in cats. Bipolar stimulating electrodes were stereotactically placed in the crossed component of the olivocochlear bundle (COCB) and the ventral nucleus of the lateral lemniscus (VNLL). Evoked potential measurements were taken from the round window in paralyzed (Flaxedil) cats both with and without cutting the middle-ear muscles. Auditory stimuli included both clicks and tone bursts presented over a wide range of intensity levels. Biphasic stimulation of either COCB or VNLL produced effects on auditory evoked potentials typical of efferent activation, including decrements in amplitude, shifts in latency, and increased cochlear microphonic output. These effects were greatest at lower intensities. Middle-ear muscle activation did not contribute to these effects. The results will be discussed in relation to central control of peripheral sensory processing.

10:35

**Q8. Detection of amplitude modulated noise by parakeets.** R. Dooling and M. Searcy (Field Research Center, Rockefeller University, Millbrook, NY 12545)

The threshold for amplitude modulated broad-band noise was measured in parakeets using avoidance conditioning and a modified method of limits procedure. Amplitude modulation thresholds were measured at ten frequencies between 5 and 2560 Hz. Below 40 Hz, modulation thresholds were constant at about 9% modulation depth. Above 80 Hz, the modulation threshold increased with modulation frequency at the rate of about 4 dB/octave. The 3-dB down point of this function is at approximately 150 Hz. These results are compared with data obtained from human listeners using the same experimental apparatus. [Work supported by NIMH Grant #MH31165.]

WEDNESDAY MORNING, 28 NOVEMBER 1979 GRAND BALLROOM III, 8:30 TO 10:50 A.M.

### Session R. Speech Communication III: Identification and Recognition of Speech by Machines

June E. Shoup, Chairman

*Speech Communications Research Laboratory, 806 West Adams Boulevard,  
Los Angeles, California 90007*

Chairman's Introduction—8:30

#### *Contributed Papers*

8:35

**R1. Automatic language identification.** K. P. Li and T. J. Edwards (TRW Defense and Space Systems Group, One Space Park, Redondo Beach, CA 90278)

A study on automatic language identification has recently been completed. The study included 200 talkers from ten different languages. Data from five languages were comprised of reading data from all male talkers while the other five languages were comprised of two-way conversational data from both male and female talkers. All data were band-limited to 350–3000 Hz and digitized at 8 kHz. Using an automatic segmentation program developed at TRW [J. Acoust. Soc. Am. Suppl. 1 **64**, S179(A) (1978)] which provided a set of six acoustic-phonetic labels, several forms of Markov types of models and classification methods were developed to identify each language independently of the talker and context. Half of the

talkers were used for training the models, while the other half was used for independent test. Classification methods, statistics of the data, and identification results will be presented.

8:50

**R2. Performance trade-offs in dynamic time warping algorithms for isolated word recognition.** C. S. Myers, L. R. Rabiner, and A. E. Rosenberg (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

The technique of dynamic programming for time registration of a reference and a test utterance has found widespread use in the area of discrete word recognition. Recently a number of variations on the basic time warping algorithms have been proposed by Sakoe and Chiba, and Rabiner, Rosenberg, and Levinson. These

algorithms all assume the test input is an isolated word whose endpoints are known (at least approximately). The major difference in the methods are the global path constraints (i.e., the region of possible paths), the local continuity constraints on the path, and the distance weighting and normalization used to give the overall minimum distance. The purpose of this investigation is to study the effects of such variations on the performance of different algorithms for a realistic speech data base. The performance index is based on speed of operation, memory requirements, and recognition accuracy of the algorithm. Preliminary results indicate, in most cases, only small differences in performance among the various methods.

9:05

**R3. On the use of clustering for speaker-dependent isolated word recognition.** L. R. Rabiner and J. G. Wilpon (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

Speaker-trained, isolated word recognizers have achieved notable success in a wide variety of applications. The training for such systems generally involves a single (or sometimes two) replication(s) of each word of the vocabulary by the designated talker. Word reference templates are then formed directly from these replications. In recent work on speaker-independent word recognition, it has been shown that statistical clustering procedures provided an effective way for determining the structure in multiple replications of a word by different talkers. Such techniques were then used to provide a set of reference templates based on the clustering results. In this talk we discuss the application of clustering techniques to speaker-trained word recognizers. It is shown that significant improvements in recognition accuracy are obtained when using templates obtained from a clustering analysis of multiple replications of a word by the designated talker. It is also shown that recognition accuracy did not change with time (over a 6-month period) for any of the subjects tested, thereby indicating that the reference templates were reasonably stable.

9:20

**R4. New techniques for automatic speaker verification using telephone speech.** S. Furui<sup>a1</sup> (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

This paper describes new techniques for automatic speaker verification using telephone speech. The operation of the system is based on a set of functions of time obtained from acoustic analysis of a fixed, sentence-long utterance. These time functions are expanded by orthogonal polynomial representations and compared with stored reference functions. After dynamic time warping, a decision is made to accept or reject an identity claim. Three sets of experimental utterances were used for the evaluation of the system. The first and second sets each comprises 50 utterances by 10 customers each and a single utterance by 40 imposters recorded over a conventional telephone connection. The third set comprises 26 utterances by 21 customers each and a single utterance by 55 imposters recorded over a high quality microphone. The first and third sets were uttered by male speakers, whereas the second set was uttered by female speakers. Reference functions and decision thresholds were updated for each customer. The evaluation indicated mean error rates of 0.19%, 0.36%, and 0.77% for each utterance set, respectively.

<sup>a1</sup> Permanent address: Electrical Communication Laboratories, Nippon Telegraph and Telephone Public Corporation, Musashino, Tokyo, Japan.

9:35

**R5. A conversational-mode airline information and reservation system using speech input and output.** S. E. Levinson and K. L. Shipley

(Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

We describe a conversational-mode speech understanding system which enables its user to make airline reservations and obtain timetable information through a spoken dialog. The system is structured as a three level hierarchy consisting of an acoustic word recognizer, a syntax analyzer, and a semantic processor. The semantic level controls an audio response system making two-way speech communication possible. The system is highly robust and operates on line in a few times real time on a laboratory minicomputer. The speech communication channel is a standard telephone set connected to the computer by an ordinary dialed-up line.

9:50

**R6. Automatic acoustic-phonetic segmentation using a hidden Markov model.** T. J. Edwards and K. P. Li (TRW Defense and Space Systems Group, One Space Park, Redondo Beach, CA 90278)

In a previous meeting [J. Acoust. Soc. Am. Suppl. 1 **64**, S179(A) (1978)], we presented an evaluation of TRW's automatic segmentation program. More recently, we have sought to improve segmentation performance using a hidden Markov model (HMM) to predict the underlying phonetic segments. Utilizing a hand-transcribed data base, the HMM was initially directly calculated on the segmentation output and transcription and the new segmentation recognition result evaluated. Using an iterative procedure [J. Baker, in *Speech Recognition*, edited by R. Reddy (Academic, New York, 1975), pp. 521-542] the HMM was then trained using only the segmentation output and compared with the previously obtained segmentation result. Using different HMM initial conditions, the model was again trained in a test for convergence. A comparison of each of these HMM for segmentation will be provided and evaluation of the models will be discussed.

10:05

**R7. Application of post-correction techniques to connected digit recognition.** L. R. Rabiner and C. E. Schmidt (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

A scheme is proposed for connected digit recognition in which a set of isolated word templates is used as reference patterns and an unconstrained dynamic time warping algorithm is used to literally "spot" the digits in the string. The recognizer keeps track of a set of candidate digit strings for each test string. The string with the smallest accumulated distance is used as a preliminary string estimate. To help improve the recognition accuracy, we have considered two "post-correction" techniques applied to the entire set of hypothesized digit strings. One technique creates a reference string by concatenating reference contours of the digits of the string and compares this to the test string using a constrained dynamic warp algorithm. The other technique performs a similar comparison using voiced-unvoiced-silence contours instead of the measured features. Small but consistent improvements in recognition accuracy have been obtained using these techniques for both speaker-trained and speaker-independent systems over dialed-up telephone lines.

10:20

**R8. A statistical approach to metrics for word and syllable recognition.** Melvyn J. Hunt (Bell-Northern Research, 3 Place du Commerce, Verdun, Quebec, Canada, H3E 1H6)

Time-warping pattern-comparison algorithms are widely used in speech recognition. Two words or syllables being compared are described by a series of time frames each containing values of a set of acoustic parameters. After time alignment, the squared distance between the patterns is summed over the parameters within a frame and then across frames. The sum obtained is assumed to be proportional to the log probability of the two patterns having the



same identity. This assumption is generally invalid, but it may be made substantially true by analyzing the variability between different examples of the same syllable and adjusting the metric accordingly. Variability is estimated both as a function of frame position within the syllable as a function of the acoustic parameters. In the latter case, within- and between-class covariance matrices can be estimated and standard linear discriminant analysis methods applied. This permits the combination of disparate acoustic parameters into a single distance measure. In particular, combining frame and frame-difference parameters allows one to use time development information and to take inter-frame correlations into account.

10:35

**R9. Synthesis-based recognition of continuous speech.** K. K. Paliwal and P. V. S. Rao (Speech and Digital Systems Group, Tata Institute of Fundamental Research, Homi Bhabha Road, Bombay 400 005, India)

An acoustic phonemic recognition system for continuous speech is presented. The system utilizes both context-dependent and context-independent characteristics of the speech signal to achieve recognition. The interphonemic contextual effects contained in formant transitions are incorporated into the system by using the synthesis-based recognition approach of Thosar and Rao [IEEE Trans. Acoust. Speech Signal Proc. ASSP-24, 194-196 (1976)]. It is shown that the information contained in transition segments of the speech signal improves the performance of the system considerably. Recognition of continuous speech is accomplished here in three stages: segmentation, steady-state recognition and synthesis-based recognition. The system has been tried out on 40 test utterances, each 3-4 s in duration, spoken by a single male speaker and the following results are obtained: 5.4% missed segment error, 8.3% extra segment error, 52.3% correct recognition using only steady-state segments and 62.0% correct recognition using both steady-state and transition segments.

WEDNESDAY MORNING, 28 NOVEMBER 1979 GRAND BALLROOM II, 11:15 A.M. TO 12:05 P.M.

## Plenary Session

Harvey Fletcher, Chairman

*Department of Physics and Astronomy, Brigham Young University, Provo, Utah 84602*

### Chairman's Introduction—11:15

**Fifty years of musical acoustics in the Acoustical Society of America.** Robert W. Young (Naval Ocean Systems Center, San Diego, CA 92152)

Many of the organizers of the Acoustical Society of America in 1929 included musical acoustics among their broad acoustical interests. The first President, Harvey Fletcher, included chapters on the sounds of musical instruments and the perception of music in his *Speech and Hearing* published in 1929; eighteen years later he outlined an Institute of Musical Acoustics "designed to provide a scientific basis and techniques whereby music may contribute more fully to the aesthetic life of a larger number of people." The second President was Dayton C. Miller, well known as the author of *The Science of Musical Sounds*, which in 1929 was in its second edition; he had exerted a strong influence in widening the scope of the Acoustical Society of America from architectural acoustics to *all* acoustics including musical acoustics. Vern O. Knudsen was the third President of the Society; in 1939 he predicted "recent discoveries in the nature of music and hearing, and especially developments in electromusical instruments, will mark the beginning of the world's greatest era in music." The Technical Committees, which are now so valuable in the Society, grew out of a suggestion by G. W. Stewart in 1940 that a special committee be formed to give those members of the Society interested in musical acoustics "an outlet of expression of opinion which would be more effective than if from one individual." Much of musical acoustics in these first 50 years of the Society may be described as studies of transient modifications of tone quality including vibrato, and studies of rigid scales and departures from rigid scales. Since two decades ago, research discloses evidence of inharmonic components in the transient sounds of every traditional musical instrument including the voice; variability in intonation is typical of almost all instruments by use of vibrato, or strings or pipes not tuned in unison. Gratifying progress has been made particularly in the past two decades in theory and experiment for the vibrations of string, brass, and woodwind instruments. The reviews of patents on electrical musical instruments, written for the Journal by Daniel W. Martin for more than 30 years, collectively document the remarkable "developments in electromusical instruments" that Knudsen predicted "will mark the beginning of the world's greatest era in music."

**Mini-session on lecture–demonstrations in acoustics**

Henry E. Bass, Chairman

*Department of Physics and Astronomy, University of Mississippi, University, Mississippi 38677*

**Chairman's Introduction—12:15**

***Invited Paper***

Following presentation of the paper the psychoacoustics cassette tapes prepared at Harvard under the direction of David Green will be available for listening.

**12:20**

**Oscilloscope demonstrations for musical acoustics.** R. Dean Ayers (Department of Physics–Astronomy, California State University, Long Beach, CA 90840)

A series of lecture–demonstrations utilizing a display oscilloscope has been prepared on a video cassette. Topics covered include: electronic spirograph patterns, use of the oscilloscope, waveforms and tones, addition of harmonics, damped oscillations, resonance and transients, Fourier analysis, speech synthesis, the beat phenomenon, and nonlinear effects. Some of these topics are made more concrete for students by the assignment of related graphical homework exercises; examples of these will be presented.

WEDNESDAY AFTERNOON, 28 NOVEMBER 1979 BONNEVILLE ROOM 2, 2:00 TO 4:55 P.M.

**Session S. Architectural Acoustics II and Psychological Acoustics III: Consideration of the Sensorially Handicapped in the Design of Interior Spaces**

David Lubman, Chairman

*Hughes Aircraft Company, Building 600 MS E-235, Fullerton, California 92634*

**Chairman's Introduction—2:00**

***Invited Papers***

**2:05**

**S0. Chairman's introductory remarks.** David Lubman (Hughes Aircraft Co., Building 600, MS E-235, Fullerton, CA 92634)

The recent trend toward mainstreaming the handicapped, implies that acoustical designers must now consider the needs of persons with hearing and visual handicaps. What are these needs, and how can designers meet them? Invited speakers and panelists have been assembled to address these questions. They will review evidence showing that sensorially handicapped persons have an enormous stake in the acoustical quality of interior spaces intended for work, learning, and recreation. Examples of considerate acoustical design will illustrate how the ability of the hearing impaired to understand speech, and the mobility of the blind, can be greatly enhanced. Finally, a panel of invited speakers and other experts will consider whether the prevailing standards of acoustical acceptability are adequate for sensorially handicapped persons.

**2:15**

**S1. Interpretations of speech and noise characteristics of NTID learning centers.** J. C. Webster (Department of Communication Research, National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, NY 14623)

The visual and acoustic design of four types of learning centers will be discussed. Levels of background noise and teachers voice levels in four classrooms will be interpreted in terms of expected speech discrimination. A flow chart relating degrees of hearing impairment and acceptable background noise levels for successful hearing aid use will be presented for comment.

2:35

**S2. Are rooms with "good acoustics" really good for listeners with impaired hearing?** Anna K. Nabelek (Department of Audiology and Speech Pathology, University of Tennessee, TN 37916)

The following criteria used presently for designing room acoustics for listeners with normal hearing will be discussed in light of special requirements for listeners with hearing impairment: (1) loudness level, (2) reverberation times, (3) diffuse sound, (4) background noise, and (5) masking noise as used in open-plan offices. Needs and possible remedies for hard of hearing listeners, and for listeners with hearing impairment who use hearing aids, will be brought into attention. [Work supported by NIH.]

2:55

**S3. Open-plan offices may be hazardous to persons with impaired hearing.** B. Leshowitz (Department of Psychology, Arizona State University, Tempe, AZ 85281)

The increasingly popular open-plan design of office space defines work zones not by room partitions, but by free-standing barriers and specially constructed acoustical environments. The key ingredient in achieving privacy, while maintaining the desired flexibility of the open-plan office, is the insertion of low-level masking noise throughout the space. While the level chosen for the background noise depends on several architectural and personal factors, broadband noise at levels up to 52 dBA is acceptable, at least to most people with normal hearing (ASTM, 1976). But what about the substantial segment of the population that is known to have a significant hearing problem? It is well known, for example, that a major limitation of the listener with sensorineural hearing damage is the inability to understand audible speech presented against a background of competing sound. For the impaired listener, the ratio of speech to background level (or S/N) must be increased by about 10 dB over that measured for normally hearing individuals. It thus appears that the modern design of office space, emphasizing acoustical sound insulation, is totally incompatible with the auditory abilities of many potential, especially elderly, users of these enclosures. In the present paper the limitations of the open office plan for handicapped listeners are examined; where appropriate, design criteria are recommended.

3:15

**S4. Open-plan offices pose a disadvantage to hearing-impaired people.** A. H. Suter (U.S. Department of Labor-OSHA), Washington, DC 20210)

Theoretically, open-plan offices are designed so that their occupants may study without being distracted, as well as converse with each other and on the telephone. Design criteria using masking noise are based on assumptions that talkers, listeners, and listening conditions are "average." These assumptions are reviewed, especially as they affect hearing impaired people. There is evidence that noise levels of 47-50 dBA can degrade communication of the hearing impaired without affecting those who hear normally. A study of speech discrimination in noise showed that even people with mild hearing losses had considerably more difficulty understanding speech than their normal-hearing counterparts. Using predictions based on the ANSI standard for calculating the articulation index, a mean difference in AI units of up to 0.15 was found between discrimination scores of mildly hearing impaired and normal-hearing groups for the same conditions. This value translates to a speech-to-noise ratio of approximately 4.5 dB, given similar spectra for speech and noise. Negative speech-to-noise ratios may occur, especially when talkers lower their voices out of courtesy to neighbors or because privacy is desired, thus exacerbating the disadvantage of the hearing impaired.

3:35

**S5. Consumer needs of the hearing impaired: The European experience and its applicability to North America.** Henry Tobin (Consumers Organization for the Hearing Impaired, P.O. Box 166, Owings Mills, MD 21117)

Several European nations have established a practice of building electronic aids for hearing-impaired listeners into public meeting rooms, churches, and theatres. The acceptance of these systems by their users and by the various public and private agencies is assessed. The provision of such listening systems is another way for acoustical designers to give consideration to the hearing handicapped. The applicability of the European experience to North America is discussed.

3:55

**Panel Discussion**

Session T. Engineering Acoustics IV: Transducers

Gordon E. Martin, Chairman

Naval Ocean Systems Center, Code 724, San Diego, California 92152

Chairman's Introduction—2:00

Contributed Papers

2:05

**T1. Reciprocity calibration of transducers in a plane wave resonator.** Steven Garrett<sup>a)</sup> (University of Sussex, School of Mathematical and Physical Sciences. Falmer, Brighton, England, BN1 9QH)

The recently published expression for transducer calibration in a plane wave resonator by the reciprocity method [I. Rudnick, *J. Acoust. Soc. Am.* **63**, 1923 (1978)] has been tested against a Rayleigh disk. The resonator tube was terminated at one end by a dual electret transducer and at the other end by an electro-dynamic transducer. All three transducers were reversible allowing three independent reciprocating calibrations which agreed to better than 1% (0.1 dB). The deflection of the disk was measured by monitoring the change in inductance of a coil surrounding the resonator and the disk. The disk angle detection system was self-calibrating and permitted absolute measurement of the flow velocity in the vicinity of the disk when the moment of inertia of the disk and damping coil were known. [Research supported by the F. V. Hunt Postdoctoral Fellowship of the Acoustical Society of America with funds provided by the Office of Naval Research.]

<sup>a)</sup> Present address: Department of Physics, University of California, Berkeley, CA 94720.

2:20

**T2. Cross-spectral method of measuring acoustical intensity—correcting phase mismatch error by calibrating two microphone systems.** G. Krishnappa (Engine Laboratory, Division of Mechanical Engineering, National Research Council of Canada, Ottawa, Canada K1A 0R6)

The accuracy of measuring acoustic intensity using two closely spaced microphones by correcting the phase mismatch error by measuring the transfer function between the two microphone systems is examined. The two measuring microphone instrumentation systems were calibrated by flush mounting the two microphones in a rigid circular plate attached to the end of a long length of circular pipe. A random noise source was mounted at the other end of the piping system. The two microphones were exposed to the same noise (phase and pressure levels) over a wide range of frequencies. The accuracy of the measurement method was verified by creating a sound field in an anechoic room by a loudspeaker and generating plane-wave propagation inside a long length of pipe with an anechoic termination. The measurement accuracies were very satisfactory. This method has the advantage of eliminating the recording and processing of two sets of data required in circuit switching technique.

2:35

**T3. Carrier-operated microphone preamplifier with self-testing capability.** Edith L. R. Corliss and William B. Penzes (National Bureau of Standards, Washington, DC 20234)

In carrier-operated preamplifiers, a condenser microphone can be placed in series with an inductor in one arm of a modified Van Zelst

bridge circuit [M. D. Burkhard *et al.*, *J. Acoust. Soc. Am.* **32**, 501–504 (1960)]. The microphone capacitance and the inductor are resonant at the carrier frequency. Consequently, the impedance of this arm is relatively low, and the signal-to-noise ratio can exceed that of a conventional high-impedance preamplifier circuit by an order of magnitude or more. Sound-induced motion of the diaphragm modulates the carrier frequency; demodulation and amplification produce the output signal. Advances in microelectronic components since 1960 include the availability of temperature-stable integrated circuits, small crystal-controlled oscillators, and temperature-compensated voltage references. These components were used to construct a carrier preamplifier of small size suitable for use with "half-inch" microphones. Since the bridge circuit balance is dependent upon the electrical impedance of the microphone, changes in this impedance associated with changes in microphone sensitivity can be detected by use of an audio-frequency insert voltage. This insert technique provides a simple and convenient *in situ* check of the microphone and preamplifier. Tests for preamplifier battery supply and polarization voltage are also provided.

2:50

**T4. Transient response optimization of piezoelectric disk transducers by time domain techniques.** H. A. Wolf and J. M. Lawther (Applied Research Laboratory, The Pennsylvania State University, State College, PA 16801)

The goals of this research have been to find and demonstrate design procedures for reducing the mechanical ringing of piezoelectric disk transducers under impulsive electrical excitation. The primary criterion of the optimization procedure is the minimization of face velocity ring-down peaks under the constraint of limiting loss from the peak velocity. Improvements to the transient response have been sought through the determination of appropriate matching networks at the two mechanical ports and at the electrical port. The optimization is accomplished by performing a gradient search over a parameter space defining the port matching configurations. The transducer and front-side mechanical matching sections are modeled with distributed parameters. Back-side loading is restricted to being resistive. Conventional inductance-tuned matching at the electrical port is demonstrated to be detrimental in reducing transient ringing. The findings indicate a natural division of the optimization process. Lightly backed and heavily backed transducers define the categories, each with an appropriate set of design criteria. Within each category, these new time domain optimization procedures result in improved transient performance over previously reported techniques based on frequency domain analysis. Experimental demonstrations are provided to verify the theoretical design improvements.

3:05

**T5. Advances in fiber interferometer hydrophones.** J. H. Cole and J. A. Bucaro (Naval Research Laboratory, Washington, DC 20375)

Progress on the development of fiber interferometer hydrophones is presented. Baseline low-frequency noise has been systematically

reduced over the past year. A dramatic improvement in minimum detectable pressure of fiber hydrophones at frequencies between 200 Hz and 1 kHz was recently achieved by the incorporation of state of the art optical components. The potential for increased performance of fiber interferometers is discussed.

3:20

**T6. Pressure sensitivity amplification in interferometric fiber optic hydrophones.** R. Hughes and J. Jarzynski (Naval Research Laboratory, Washington, DC 20375)

The induced optical phase change when a pressure is applied to a fiber optic hydrophone is examined both theoretically and experimentally. The induced phase change is calculated both with and without a plastic jacket around the glass fiber. It is shown that the phase change predicted from a detailed three-dimensional analysis can be adequately described in terms of the much simpler two-dimensional generalized plane strain approximation. Sensitivities calculated assuming hydrostatic boundary conditions are shown to correspond closely in value with static pressure-sensitivity measurements. The application of a plastic jacket can substantially increase the pressure sensitivity of the fiber. The increased phase shift is due primarily to the tendency of the jacket to increase the axial strain experienced by the glass fiber. It is predicted that maximum pressure sensitivity (up to 30 times that of the bare fiber) will be achieved by jacketing with Teflon.

3:35

**T7. Silicon rubber for optical hydrophones.** Peter Shajenko and James Flatley (Naval Underwater Systems Center, New London, CT 06320)

Elasto-optic properties of materials can be used for modulating the laser beam by the acoustic pressure. The compressibility and the change of index of refraction with pressure of silicon rubber were measured using the technique of interferometry, and the results were applied to determine the voltage sensitivity and minimum detectable pressure of an optical hydrophone. Both values compare favorably with those of a conventional piezoelectric hydrophone.

3:50

**T8. Studies of PVF<sub>2</sub> transducers.** E. F. Carome (John Carroll University, Cleveland, OH 44118)

Thin film polyvinylidene fluoride (PVF<sub>2</sub>) polymer transducers are being used for various applications over the frequency range 0.2 to 30 MHz. Operated in the thickness mode, they have been employed as sources and detectors of longitudinal and surface waves. Wide-band impulses are easily generated and detected with these elements, so that a transducer's performance can easily be evaluated by examining its direct and Fourier transformed response. Detailed data are presented on the sensitivity, frequency response, and other characteristics of various PVF<sub>2</sub> transducer configurations. [Work was partially performed at Stanford University and supported by ONR.]

4:05

**T9. Utilization of semiconductor strain gauges in pressure gradient sensor designs.** F. W. Cuomo and D. J. Hilliker (Department of Physics, University of Rhode Island, Kingston, RI 02881 and Naval Underwater Systems Center, Newport, RI 02840)

The performance of a pressure gradient sensor of the fixed baffle type embodying semiconductor strain gauges is discussed. The output is optimized by the utilization of a bridge circuit providing good linearity and temperature compensation. Experimental data are obtained by means of a low-frequency acoustic calibrator similar in design to an earlier device [T. H. Ensign, *J. Acoust. Soc. Am.* **50**,

108(A) (1971)] and compared with identical gradient transducers employing ceramic elements.

4:20

**T10. Optimization study of a parametric hydrophone with a slow waveguide.** J. Jarzynski and R. D. Corsaro (Naval Research Laboratory, Washington, DC 20375)

The performance of a parametric hydrophone with a cylindrical slow waveguide between the pump and receiver transducers was calculated as a function of the waveguide dimensions and material properties. The computer program used is described by Corsaro and Jarzynski [*J. Acoust. Soc. Am.* **66**, 895-904 (1979)]. The parametric conversion gain and beam patterns were calculated for waveguide lengths in the range 0.1 to  $9\lambda$ , where  $\lambda$  is the low-frequency wavelength in water; for waveguide radius/length ratios in the range 0.03 to 0.5; and for waveguide materials with sound speeds in the range 400-1500 m/s. The general conclusion is that large gains in parametric conversion efficiency ( $\sim 20$  dB) and in directivity are achievable for geometries less than a few wavelengths long, with diameter/length ratios no less than 0.1, and with material sound speeds in the range 800-1200 m/s. With longer or thinner cylinders, or with materials having speeds less than 800 m/s, the side-lobe levels increase, and both efficiency and directivity are typically reduced. These effects are predictable from Jacobi's analysis of sound propagation in slow waveguides [*J. Acoust. Soc. Am.* **21**, 120 (1949)].

4:35

**T11. Abstract withdrawn.**

4:50

**T12. The motional impedance of an electro-dynamic loudspeaker.** F. M. Murray (James B. Lansing Sound, Inc., 8500 Balboa Boulevard, Northridge, CA 91329)

For many years the electrical equivalent of the motional impedance of an electrodynamic loudspeaker has been represented as a lumped parallel resonant circuit with the tacit understanding that circuit elements do become distributed and that such complications are "beyond the scope of this course." Recent experiments with high efficiency compression drivers show that these distributed elements can be analyzed and may be very important; in some cases causing cutoff characteristics similar to  $M$ -derived filters rather than the classic constant- $K$  filters normally assumed. The author recently presented qualitative results (F. M. Murray and H. M. Durbin, Audio Engineering Society 63rd Technical Meeting, Paper D8) of experiments to control higher-order modes of the motional impedance of such transducers. This paper presents a quantitative analysis to

further understand this phenomenon. Some still unresolved questions will be discussed. When fully developed this new theory promises to provide better insight into gross anomalies sometimes attributed to other sources.

5:05

**T13. Ultrasonic bond evaluation in multilayered media.** R. M. Havira (Schlumberger-Doll Research, P.O. Box 307, Ridgefield, CT 06877)

This paper discusses a spectroscopic and time domain method for the evaluation of bonding and the determination of thickness in multilayered media using broadband ultrasonics. Mathematical modeling of multilayered media based on Z-transform and scattering matrix techniques are presented. Experimental results using 500 KHz pulses are presented for water-steel-cement boundaries, water-steel-water boundaries and for a fluid-filled delamination between the steel and cement. The comparison between experimental and theoretical results are discussed.

WEDNESDAY AFTERNOON, 28 NOVEMBER 1979 BONNEVILLE ROOM 1, 2:00 TO 5:05 P.M.

### Session U. Musical Acoustics III: Electronics in Music

Alan C. Ashton, Chairman

*Department of Computer Science, Brigham Young University, Provo, Utah 84602*

**Chairman's Introduction—2:00**

#### *Invited Papers*

2:05

**U1. Serial-parallel architecture for real-time synthesis of music.** E. Ferretti (Department of Computer Science, University of Utah, Salt Lake City, UT 84112)

Parallel architecture is extremely suitable for the synthesis of music. For example the performance of several instruments at the same time is a directly related task for parallel architecture. Serial architecture on the other hand is ideally suited for optimizing real-time tasks. The reasons for serial-parallel architecture are (1) to utilize a single-type module for computing all tasks in real time, (2) to have a means of adjusting the number of tasks without changing the basic system or violating real-time requirements, and (3) to have a means of computing time waveforms which require a large amount of transcendental functions for approximating natural sounds. This paper will discuss the issues of system configuration, accuracy, dynamic range of output versus computation, multiple instruments, and modeling of natural sounds for computer music. Some examples of computer music will be presented, and a high-speed film of a vibrating rubber band will be shown. [This work was supported in part by a private grant from Michael Lay, and use was made of computer facilities in the Computer Science Department.]

2:35

**U2. Automatic music performance with computers.** A. C. Ashton and R. F. Bennion (Brigham Young University, UT 84602)

Developments of the Utah-BYU music project which have culminated in the design and implementation of a portable computer-driven music-generating system capable of interpretive playing of transcribed musical scores will be discussed, and a film depicting concurrent playing and graphing of music will be shown. The portable music system will be used to demonstrate direct interactive performance of a number of musical selections which have been notated in the Utah-BYU linear music language. Musical parameters such as pitch, duration, tempo, key, voice orchestration, and transposition can be notated for musical performance and can be dynamically modified during the automatic playing of the music. Music generation is accomplished by digital-to-analog conversion of stored digital waveforms which are programmably selectable at music generation time. Recordings of a computer-driven pipe organ will also be played and the organ-computer interface will be discussed.

3:05

**U3. Sound analysis-synthesis and interactive composition.** J. W. Beauchamp (School of Music and Department of Electrical Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801)

Two distinct projects are under way: (1) Derivation of synthesis models from the analysis of acoustic musical instrument sounds and their use with software-based synthesis programs on large computers. (2) Development of an interactive computer-controlled synthesizer system housed in the University of

Illinois Music Building. It is hoped that in the not-too-distant future these projects will be joined. Current focus in analysis is to estimate the resonant filter characteristics of a wind instrument and use this in conjunction with time-variant Fourier analysis to derive an excitation signal. This is further processed to reveal a nonlinear distortion function and an *index signal*. In the simplest case, synthesis over wide dynamic and pitch ranges is possible by merely varying the fundamental frequency and the index signal while keeping the filter and distortion parameters fixed. The interactive system uses a compositionally oriented language called PLACOMP written for the PLATO IV time-sharing system. A PLACOMP score is translated into numbers which are quickly delivered to a hard-wired synthesizer. The turnaround time between delivered completion and audible result varies between a few seconds and a few minutes depending on the complexity and length of the composition.

3:35

**U4. Dumb ways to play intelligent instruments.** M. V. Mathews (Bell Laboratories, Murray Hill, NJ 07974 and IRCAM, Paris, France)

In the last five years a new generation of digital instruments has been developed using the new integrated circuit technology. In contrast to previous digital instruments, which were confined to studios, these new instruments are small enough, cheap enough, portable enough, reliable enough, and powerful enough to form a new class of performance instruments. In addition to sound synthesis circuits, they include mini- and microcomputers with general computing power and with digital memories. Because of their computational power, we call them intelligent instruments. Musicians must learn very different performance techniques to take advantage of the potential of these instruments. In this paper we review some of the initial ways in which the instruments have been played. We conclude that although many of the performances are interesting, much must be learned before the full potential of these instruments is utilized.

### Contributed Papers

4:05

**U5. Sequential drum.** M. V. Mathews (Bell Laboratories, Murray Hill, NJ 07974)

A sequential drum is described. It consists of a square surface 20" by 20" which is hit with conventional drum sticks. Transducers on the surface generate three electric signals each time the surface is hit. One signal is a pulse whose amplitude is proportional to the impact. The other two signals encode the *x* and *y* positions of the stroke. The three signals are transmitted to a digital sound synthesizer (Alles machine) which generates a note each time a signal is received. The loudness of the sound is controlled by the amplitude pulse. The timbre of the sound is controlled by the *x* and *y* signals. The pitch of the sound is obtained from a score stored in the memory of the digital synthesizer. Each time the drum is struck, the next pitch in the score is played; thus, the desired sequence of pitches is produced automatically by the instrument.

4:20

**U6. A study of stretched harmonies.** K. T. Marcus and M. V. Mathews (Bell Laboratories, Murray Hill, NJ) and J. R. Pierce (California Institute of Technology, Pasadena, CA 91109)

A class of nonharmonic sounds was synthesized by stretching normal harmonic sounds. Stretching means uniformly expanding the logarithmic frequency ratios between the fundamental frequencies of the tones in a scale and between the fundamental of each tone and its overtones. Stretched passages were studied to see what properties are invariant to stretching. Stretched cords have the same coincidence of overtones as unstretched chords; hence, harmonic perceptions, which depend on overtone coincidence, should be invariant. By contrast, perceptions which depend on periodicity pitch or on the Rameau fundamental bass should be destroyed by stretching. A number of traditional pieces and chords were synthesized in stretched and unstretched forms. These samples were evaluated by (1) informal listening, (2) tests of the identification of the key of the material, and (3) tests of the perception of cadences. Results showed both similarities and differences between stretched and

unstretched materials. We believe stretched sounds have perceivable structures that can be utilized in musical compositions.

4:35

**U7. Musical spectroscopy I: The real-time power spectrum in instrument and voice teaching.** Charles E. Potter (The Juilliard School, New York, NY 10023) and Dale T. Teaney (IBM Watson Research Center, Yorktown Heights, NY 10598)

We have explored the uses of a real-time sound color "machine" in the music studio environment. The spectral analysis is performed by a 512 line, 33-ms FFT instrument; the output of which is sampled every 65 ms at a logarithmically uniform density of 96 points/8va and stored in one of two 512- $\times$  10-bit buffers. The 5-8va spectra are displayed as graphs one above the other. The rms scale is well suited to a single instrument. A piano keyboard graphic underlines each spectrum. In operation, a model spectrum by the teacher or from a recording is captured by action of a foot switch; the student then observes his own spectrum, running in real time or frozen, on the graph below the model. The spectrometer provides a graphic notation that relates the kinesthetics of production behavior to the acoustic properties of the instrument on the one hand and to the aural penetration of the sound on the other hand. A change of vowel, an alternate fingering, or the influence of a mute is usually enough for the student to get the idea of what the display means to him, and "the machine" then becomes a sort of recording secretary in the performance training dialog.

4:50

**U8. Musical spectroscopy II: The sound score in composition and analysis.** Robert Cogan (New England Conservatory, Boston, MA 02115) and Charles E. Potter (The Juilliard School, New York, NY 10023) and Dale T. Teaney (IBM Watson Research Center, Yorktown Heights, NY 10598)

In *Visible Speech* [R. Potter *et al.*, Van Nostrand, New York (1947)] musical applications of a suitably adapted sonogram were

foreseen, and Charles Seeger [Musical Quarterly 44, 184-195 (1958)] gave both practical examples of and the theoretical apology for descriptive music writing. In *Sonic Design* [Cogan and P. Escot, Prentice Hall, New Jersey (1976)] extended music-spectrographic methods are developed for both analytical and procedural purposes. Our spectrometer (see preceding abstract) generates a simple sound score on a grey-scale storage monitor. The display is, in effect, proportional notation of the spectral components over a 5-octave

range against a 5- to 50-s horizontal span. The spectral density function, presented at the (z axis) luminance signal, is weighted to give approximately uniform brightness for a sine tone, top to bottom. Klavier or register staff lines are overlaid, and an audio-graphic equalizer permits modest refinements. Normally read in real time while listening to the music, the sound score can be photographed frame by frame. Examples from diverse musics will be presented.

WEDNESDAY AFTERNOON, 28 NOVEMBER 1979 BONNEVILLE ROOM 5, 2:00 TO 5:05 P.M.

## Session V. Noise V: Temporal and Spatial Sampling Statistics and Techniques

Norman J. Meyer, Chairman

*Wyle Laboratories, 128 Maryland Street, El Segundo, California 90245*

**Chairman's Introduction—2:00**

### *Invited Papers*

**2:05**

**V1. Results of monitoring of noise and air quality in feeder areas at various proximities to freeways for matched groups of school children.** J. W. Swing (California Office of Noise Control, Department of Health Services, 2151 Berkeley Way, Berkeley, CA 94704)

Community noise levels throughout feeder areas serving 16 grammar schools in the Los Angeles County School District were monitored in order to determine the environmental noise exposure experienced by 3rd and 6th graders in socioeconomically matched groupings. Schools for this study were selected such that the noise environments for one-half of the feeder areas would be largely controlled by freeway traffic noise. Noise monitoring at school sites within each feeder area was also conducted. Additionally, indoor and outdoor air quality monitoring was conducted on a rotational basis during December 1978 and January 1979 at each school site. Estimates of each feeder area's relative air quality have been extrapolated from the school studies which were in turn correlated to South Coastal Air Basin air quality data. Community noise levels, based upon approximately 350 site days of monitoring, are presented along with each feeder area's socioeconomic and ethnic population descriptors. The influence of freeway proximity on environmental noise levels and ambient air quality is discussed.

**2:35**

**V2. Combining physical modeling information and statistics in prediction of airport noise in communities.** Dwight E. Bishop (Bolt Beranek and Newman Inc., P.O. Box 633, Canoga Park, CA 91305)

The prediction of noise in communities due to transportation noise sources typically utilizes deterministic information about sound sources and sound propagation combined with statistical approaches for aid in selecting measurement positions and analyzing noise data acquired over limited time periods. In determining community noise levels to a given statistical confidence level, it will generally be most efficient, in terms of time and resources, to make utmost use of physical modeling concepts and noise source information, together with information from previous studies as to the types of temporal and spatial variations that may be involved. Tradeoffs between statistical analyses and modeling calculations will depend, obviously, upon the accuracy of the modeling concepts and the extent of factors that cannot be quantified in the model. Examples of current techniques for describing the noise environment due to aircraft noise, which combine physical modeling augmented by statistical analyses of noise monitoring data, will be described.

**3:05**

**V3. Sampling of temporal fluctuations in community noise levels.** Louis C. Sutherland (Wyle Research, 128 Maryland Street, El Segundo, CA 90245)

A theoretical basis exists for defining the statistical behavior of fluctuations in community noise levels provided the latter can be described in rather ideal terms. This paper briefly reviews some aspects



of this theoretical framework and compares the statistical behavior of several sets of actual community noise measurements to the trends expected according to theory. Deviations from the theoretical trends are illustrated which are attributed to the nonideal (i.e., non-Gaussian) behavior of real community noise environments. The results help to establish guidelines for temporal sampling schemes for various types of community noise environments to achieve a desired degree of accuracy. However, additional data are called for to more accurately define the temporal and frequency character of a wider variety of types of noise environments. Suggestions are outlined as to what type of data are required and how they may be obtained. [Work was supported, in part, by EPA, Office of Noise Abatement and Control.]

3:35

**V4. Development of temporal sampling strategies for monitoring noise.** P. D. Schomer, R. D. Neathammer (U.S. Army Construction Engineering Research Laboratory, P.O. Box 4005, Champaign, IL 61820), R. E. DeVor, and W. A. Kline (Department of Industrial Engineering, University of Illinois, Urbana, IL 61801)

This paper addresses the problem of the estimation of the long term (yearly) mean of the Community Noise Equivalent Level (CNEL) or day/night average sound level ( $L_{DN}$ ). Examination of daily average noise levels (either in mean-square pressure or in decibel units) shows that while the data may be stationary with respect to mean level over a several month period, they exhibit a strong pattern of autocorrelation in which positive correlation predominates. As a result, the sample sizes required to achieve a desired level of precision in the sample-mean estimate are much larger than they would otherwise be if the data were uncorrelated serially in time. For the data examined, to obtain an estimate of the mean level within a 5-dB range ( $\pm 50\%$  of the mean in mean-square-pressure units), sample sizes in the range of 20 to 50 consecutive daily averages would be required. If the daily averages were uncorrelated in time, only 5 to 15 consecutive daily averages would be required. The data used in this study were obtained from continuous monitoring at a number of sites in the vicinity of a busy Naval Air Station. Some data obtained from a large commercial airport were also analyzed and found to have even stronger positive autocorrelation and therefore requiring even larger sample sizes for mean-value estimation.

4:05

**V5. Unmanned noise source identification via analog tape microsampling.** Hal Watson and Jim Holman (Southern Methodist University, Dallas, TX 75275)

A new method of analog tape microsampling is discussed. The method centers about a controller which turns the analog tape recorder on and off at a low sample rate at low sound levels and at a high sample rate at sound levels above a preset threshold. Results of its application to airport/aircraft/community noise environments are discussed with regard to accuracies of level statistics and source identification.

4:35

**V6. Environmental noise assessment: Single family residential spatial analysis and methodology error analysis.** George J. Putnicki (The University of Texas at Dallas, P.O. Box 688, Mail Station BE 2.2, Richardson, TX 75080)

A number of sampling techniques have recently been developed to assist municipalities in assessing and evaluating the total environment noise climate within the community and to identify the intrusive noise sources. One methodology was developed by Wyle Research, El Segundo, California for the U.S. Environmental Protection Agency and published in July 1976. The University of Texas at Dallas, Graduate Program in Environmental Sciences, has modified the Wyle methodology for use in conducting an environmental noise assessment for the City of Dallas, Texas. This publication reports on the modifications made to the Wyle method and the results of a spatial survey conducted in a spatially homogenous single family residential land use area. The purpose of the spatial analysis was to determine the number of monitoring sites required. Prior to using the modified methodology, an error analysis was conducted for the purpose of determining the accuracy of the methodology. Nine primary monitoring sites were selected for this analysis. Using commercially available digital monitoring equipment each site was monitored for four or more continuous 24-h periods and the day-night average sound levels,  $L_{dn}$ 's, measured were compared to the 24-h  $L_{dn}$ 's computed using selected 30-min periods obtained on weekdays, weeknights, weekend days, and weekend nights. From these analyses, the required number of sites was determined and the accuracy of the methodology was validated.

Session W. Physical Acoustics IV: Miscellaneous

Robert E. Apfel, Chairman

*Yale University, Mason Lab, 9 Hillhouse Avenue, New Haven, Connecticut 06520*

Chairman's Introduction—2:00

Contributed Papers

2:05

**W1. Sound propagation in colloidal suspensions in electrolytes.** M. E. Gültepe, M. A. Barrett-Gültepe, and E. Yeager (Department of Chemistry, Case Western Reserve University, Cleveland, OH 44106)

The propagation of ultrasonic waves through colloidal suspensions in electrolytes is of interest from the standpoint of understanding sound propagation in the sea and in biological systems, and as a means for study of colloidal systems. Ultrasonic velocity and absorption measurements have been carried out in suspensions of particles of various configurations, including polystyrene latex (monodispersed spheres of, e.g., 5500 Å), kaoline (sodium form, platelets), montmorillonite (sodium form, platelets), and microcrystalline cellulose (rods) over a range of suspensions concentrations at several frequencies. The velocity changes associated with the colloid occur principally because of the difference in static adiabatic compressibility of the colloid phase and bulk electrolyte. With very small particle size—high surface area system, the change in compressibility of the solvent molecules adsorbed on the colloid surfaces may also be significant. The principal factors contributing to the additional absorption are relative motion losses, thermal conductivity across the interfaces, an ionic double layer, and adsorption-desorption relaxation. Particle-particle interactions must also be taken into account in the more concentrated suspensions. [Research supported by ONR.]

2:20

**W2. Pulmonary ventilation by high-frequency oscillations.** J. J. Fredberg (Cambridge Collaborative, Inc., Cambridge, MA 02142)

Bohn *et al.* [Fed. Proc. Fed. Am. Soc. Exp. 38 (3), 951 (1979)] reported that paralyzed beagle dogs maintained normal gas exchange for six hours or more when small oscillatory flows were maintained at the airway opening (15 cc tidal volume @ 15 Hz). These tidal volumes were 25% of pulmonary dead space and, thereby, were too small to permit convective gas exchanges with pulmonary airspaces. To test the hypothesis that effective gas exchange can be achieved by enhanced axial diffusion promoted by high-frequency oscillatory flows, I have modelled that process and computed the time constant for concentration equilibration in the lung as a function of flow amplitude and frequency. Knowing the distribution of oscillatory flows within the bronchial tree [Fredberg and Moore, J. Acoust. Soc. Am. 63, 954–961 (1978)], the enhanced diffusion due to oscillatory flows [Chatwin, J. Fluid Mech 71, 513–527 (1975)] may be integrated along the bronchial tree. The results indicate that gas concentration gradients within the lung can be substantially abolished within  $10^0$  to  $10^{-2}$  s. This suggests that diffusion enhanced by small-amplitude oscillatory flows (tidal volume  $\ll$  dead space) can support effective gas exchange in the lung in the absence of convective transport (breathing). [Work supported by the Physicians' Service Incorporated Foundation, Ontario.]

2:35

**W3. Sound velocity versus temperature measurements in tissues.** Steven R. Jacobs, Douglas A. Christensen, Steven A. Rockhold, and

Steven A. Johnson (Department of Bioengineering, 2059 Merrill Engineering Building, University of Utah, Salt Lake City, UT 84112)

We have developed a sensitive interferometric technique for measuring the variation of the velocity of ultrasound in tissue as a function of temperature. The tissue samples are placed between two transducers mounted in an invar holder, and this assembly is situated in a temperature-controlled water bath. The transmitting transducer is excited continuously by a crystal-controlled oscillator at a frequency of 10 MHz. The phase of the received signal is compared with the phase of the transmitted signal by a vector voltmeter. Variations in velocity are detected as changes in the relative phase between the transmitted and received signals. The overall accuracy of the system allows resolution to within  $3^\circ$  of phase difference. This corresponds to an accuracy of 10 ppm for a transducer spacing of 10 cm. The experimental data will be used in a larger project aimed at measuring internal body temperature noninvasively with computer-aided tomography. Temperature resolution on the order of  $0.1^\circ\text{C}$  across 1 cm of tissue are expected in the final instrument.

2:50

**W4. Rectified diffusion in the presence of surfactants.** L. A. Crum (Department of Physics, Oxford, MS 38677)

Measurements are reported of the growth of air bubbles by rectified diffusion in water at 22.1 kHz. Values were obtained of both the threshold and the rate of growth of bubbles by rectified diffusion as a function of bubble radius, acoustic pressure amplitude, liquid surface tension, and equilibrium gas concentration. Good agreement with theory was obtained for normal values of the surface tension of water for both the rectified diffusion threshold and the growth rate. With addition of a surfactant, measured values of the threshold and growth rate deviate from the calculated values, the disagreement becoming considerable at large surfactant concentrations (low surface tension). Surface-wave activity of the bubble that would increase the diffusion rate by acoustic streaming was not observed at low bubble radii and is not thought to be the cause of the disagreement. A possible explanation is given in terms of the retardation of gas diffusion by the surfactant monolayer on the surface of the bubble. [Work supported by the Office of Naval Research.]

3:05

**W5. Temperature variation of the acoustic properties of laboratory sediments.** Donald J. Shirley and David W. Bell (Applied Research Laboratories, The University of Texas at Austin, TX 78712)

The compressional wave speed and attenuation and the shear wave speed and attenuation of unconsolidated artificial laboratory sediments have been measured as a function of temperature. No overburden or pore pressure in excess of that found at a few centimeters depth was applied. The compressional wave speed of the sediment was found to vary approximately as would water with a slight increase in the slope of the curve of the sediment over that of water. Shear wave speed appears to be independent of temperature

within the accuracy of the measurement. Attenuation of both types of acoustic waves also appears to be independent of temperature. [Work supported by the Office of Naval Research.]

3:20

**W6. Visualization of acoustical processes in gases consisting of many interacting particles.** Maria Heckl (Institut für Technische Akustik, Technische Universität Berlin, West Germany)

For most acoustical calculations the carrier of the sound waves (air, water) is considered to be a continuous medium. This, however, is oversimplifying as it is known a gas consists of particles which move with randomly distributed speeds and interact by many impacts. The intention of this paper is to visualize—with the help of a film—acoustical processes under these conditions. The particles are considered as totally elastic mass-points which hit together and exchange energy and momentum according to the well-known conservation laws. With this model some simple phenomena such as sound propagation, damping, and reflection will be discussed.

3:35

**W7. Nearfield of ultrasonic transducer-lens systems: Theory with Gaussian-Laguerre formulation.** Eduardo Cavanagh and Bill D. Cook (Cullen College of Engineering, University of Houston Central Campus, Houston, TX 77004)

Within the regions of validity of the Fresnel approximations, a Gaussian-Laguerre formulation permits rapid calculation of fields of transducer-lens systems. Separation of lens from the transducer is found to change significantly the structure of field. When the lens is at a focal length from the transducer, the field is found to be symmetric with respect to the focal plane. The asymmetry when the lens is not at this distance depends on the location of the lens and the ratio of the focal length to the length of the nearfield.

3:50

**W8. Nearfield of ultrasonic transducer-lens systems: Comparison of experiment and theory.** P. L. Edwards (Department of Physics, University of West Florida, Pensacola, FL 32504), Bill D. Cook (Cullen College of Engineering, University of Houston Central Campus, Houston, TX 77004), and Henry D. Dardy (Naval Research Laboratory, Washington, DC 20375)

Local acoustic pressures in the nearfield of ultrasonic transducer-lens systems were determined by measuring the scattering from a microsphere scanned through the field. The lens was placed both adjacent to and separated by a distance of one focal length from the transducer. Typical parameters were the following: frequency 5.0 MHz, transducer diameter 0.5 in., focal length 1.0 in., diameter of microsphere 0.002 in. Experimental results show remarkable comparison with the details predicted by the Gaussian-Laguerre formulation presented elsewhere in this meeting. [This research was conducted at the Naval Research Laboratory, Physical Acoustics Branch, Washington, DC.]

4:05

**W9. Studies of acousto-optic interactions in optical fibres.** E. F. Carome, K. P. Koo, and P. B. Schmidt (John Carroll University, Cleveland, OH 44118)

Studies are continuing of acoustically induced phase and intensity modulation of light beams propagating in optical fibres. A fibre

element submerged in water is irradiated with plane acoustic waves. The optical field exiting from the output end of the fibre is precisely scanned. Acoustically induced changes in the output pattern are being examined in an effort to obtain information on fibre mode-mode coupling and other phenomena. Details of the measurements and recently obtained experimental results are presented. [Work supported in part by ONR.]

4:20

**W10. Acoustic emission from leaking valves.** J. W. Dickey (David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402)

The acoustic emission associated with leakage through air, steam, hydraulic, and water valves has been investigated. Experiments were performed to determine what characteristics of the acoustic emission may be related to leak rate with the hope that instrumentation could be developed to acoustically detect and measure leakage. The spectral and temporal characteristics of leakage-associated signals taken from externally mounted detectors or valves indicate that average acoustic amplitude in selected frequency ranges provide a useful measure of valve leakage for most styles of air, steam, hydraulic, and water valves. The existence of cavitation in water and hydraulic fluid complicates leakage measurement and, for hydraulic fluid, the sum of event amplitudes multiplied by their rate of occurrence results in a value which increases with increasing leak rate. The data collection and analysis used to support the conclusions are discussed.

4:35

**W11. A versatile fast scanner for ultrasound temperature tomography and wide aperture synthetic focus imaging.** Steven A. Johnson (Department of Bioengineering, University of Utah, Salt Lake City, UT 84112), Allen R. Grahm, Charles D. Baker (UBTL Division of the University of Utah Research Institute, University of Utah, Salt Lake City, UT 84112), George Randall (Department of Computer Science, University of Utah, Salt Lake City, UT 84112), Douglas A. Christensen, Steven R. Jacobs, and Michael J. Berggren (Department of Bioengineering, University of Utah, Salt Lake City, UT 84112)

A versatile laboratory scanner of modular design is described which can support a wide variety of ultrasound transducers and transducer arrays. The basic design consists of a rotating platform on which are mounted various types of interchangeable, temperature regulated water tanks. The platform is rotated in 0.02° steps by a high torque stepping motor. Various transmitter and receiver geometries or configurations are arranged in each water tank. One tank configuration uses spherical or cylindrical acoustic waves and a curvilinear, circular receiver array for high speed (1 to 10 s) data collection for reconstruction of tissue temperature. This method makes use of acoustic refractive index reconstruction tomography and the temperature dependence of the acoustic refractive index. This tank configuration also permits the use of microwave or high-power, focused ultrasound beams for tissue heating while simultaneously reconstructing the temperature of the tissue samples. A second tank configuration uses plane waves and a rectilinearly scanned receiving array for diffraction tomography studies. A third tank configuration permits the use of a new high resolution reflection or scattering imaging technique now known as either "synthetic focus imaging" or "reflection tomography." [Supported by NIH grants NCI IR01 CA 23430 and H2-00170 and PDP-110 from American Cancer Society.]

**Session X. Physiological Acoustics III: Cochlea, Auditory Nerve, and Models (Poster Session)**

Eric Javel, Chairman

*Boys Town Institute for Communication Disorders, Omaha, Nebraska 68131*

Chairman's Introduction—2:00

**Contributed Papers**

**X1. The role of potassium and sodium in cochlear transduction: A study with amiloride and tetraethylammonium.** A. N. Salt and T. Konishi (Laboratory of Environmental Biophysics, National Institute of Environmental Health Sciences, Post Office Box 12233, Research Triangle Park, NC 27709)

The effects of amiloride and tetraethylammonium (TEA) on cochlear microphonics (CM), action potentials (AP), and endocochlear potential (EP) were studied in guinea pigs. It has been reported that amiloride selectively reduces sodium permeability in a variety of ion transporting epithelia. Perilymphatic perfusion ( $10^{-3}$  M) or intravenous administration (20 mg/kg) of amiloride suppressed AP, but did not significantly alter CM or EP. Application of amiloride to the endolymph by iontophoretic or perfusion techniques also produced no greater changes of CM and EP than were observed during control procedures. TEA has been shown to suppress the potassium permeability increase in excitable cell membranes. Perilymphatic perfusion of TEA ( $10^{-2}$  M) did not suppress CM or EP, but AP was reduced. Iontophoretic application of TEA to the endolymph markedly suppressed CM, while EP increased in value. The effects of TEA applied to the endolymph were similar to those effects previously reported to occur during noise exposure (J. Acoust. Soc. Am. Suppl. 1 64, S132(A) (1978)). This result adds support to the hypothesis that noise induced CM suppression is a consequence of a potassium permeability decrease of the endolymph/perilymph barrier. The contribution of sodium ion movements to cochlear transduction appears less certain in the light of the insensitivity of CM and EP to amiloride.

**X2. Maintenance of cochlear function with synthetic blood and effects of potassium deficiency.** R. Thalmann, J. Kambayashi, J. Wada, J. E. DeMott, and D. Marcus (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

We have recently described a method for the vascular perfusion of the guinea pig cochlea with synthetic blood, using the fluorocarbon FC 47 as oxygen carrier [Wada *et al.*, Laryngoscope (1979) (to be published)]. We have now determined that the endolymphatic potential (EP) and the cochlear microphonics can be maintained at normal or near normal levels for perfusion periods exceeding 300 and 150 min, respectively. However, the compound action potential ( $N_1$ ) cannot be maintained for longer than 60 min. This finding may be of considerable significance for blood substitute technology in general, since  $N_1$  could be the most sensitive indicator of adverse effects of synthetic blood known to date. In other experiments we found that the EP can be maintained at normal levels for  $25.3 \pm 6.9$  (SD) minutes by  $K^+$ -free vascular perfusion. By contrast, the EP starts to decline immediately if perilymph is replaced by  $K^+$ -free media [Marcus *et al.*, J. Acoust. Soc. Am. Suppl. 1 65, S13(A) (1979)]. The possible implications of these findings with regard to the mode of generation of the EP will be discussed. [Supported by the NIH and NSF.]

**X3. Effects of substrate free vascular perfusion of the cochlea.** J. Kambayashi, J. E. DeMott, T. H. Comegys, I. Thalmann, N. Y.

Marcus, and R. Thalmann (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

During vascular perfusion of the cochlea with substrate free synthetic blood, the endolymphatic potential (EP) is maintained for  $85 \pm 20$  (SD) minutes. The subsequent decline of the EP can be prevented by 0.8 mM glucose or by 11 mM lactate, but not by succinate (although 20 mM succinate effects a partial recovery of the EP when it has declined by 40 mV). Significant breakdown of strial glycogen occurs only when the EP starts to decline. Glycogen stores in the stria are virtually depleted when the EP has dropped to about 30 mV. At about the same time, the further decline of the EP is arrested, presumably due to activation of a new substrate pool. Only following a stabilization of some 30 min does the EP resume its decline. It appears that the prolonged initial maintenance of the EP during substrate free vascular perfusion is due to the utilization of glucose from cochlear fluids, since the EP starts to decline immediately when the perilymphatic glucose pool is removed simultaneously by perilymphatic perfusion. [Supported by NIH and NSF.]

**X4. Levels of putative neurotransmitters, glutamate and aspartate, in cochlear fluids and cell layers of organ of Corti.** I. Thalmann, R. Thalmann, D. C. Marcus, and T. H. Comegys (Department of Otolaryngology, Washington University, St. Louis, MO 63110)

Glutamate and/or aspartate have been proposed as transmitters in acoustico-lateralis systems. We analyzed these substances in cochlear fluids of guinea pigs and in cell layers of organ of Corti of chinchillas. Fluids were collected with utmost care to avoid contamination. Concentrations of aspartate were  $2.6 \pm 1.2$  (SD) and  $16.3 \pm 3.5$   $\mu$ M in perilymph and endolymph, respectively. Corresponding values for glutamate were  $11.9 \pm 5.0$  and  $74.0 \pm 6.5$   $\mu$ M. Aspartate exhibited a gradient in whole organ of Corti [52 units (mmoles/kg dry weight) in third versus 109 units in basal turn]. No specific accumulations were seen in inner or outer hair cell layers; in fact, concentrations were highest in Hensen and Claudius cells. Glutamate exhibited only a slight gradient in organ of Corti (60 units in basal versus 81 units in third turn). Levels were similar in outer hair cell layer and supporting cells, but significantly lower in inner hair cell layer. No definite conclusions about a possible transmitter role of either substance can be drawn from these data. [Supported by NIH and NSF.]

**X5. Stimulus-induced release of endogenous amino acids from the *Xenopus laevis* lateral-line organ.** S. C. Bledsoe, Jr., R. P. Bobbin, R. Thalmann, and I. Thalmann (Kresge Hearing Research Laboratory, Department of Otorhinolaryngology, Louisiana State University Medical Center, New Orleans, LA 70119 and Washington University School of Medicine, Department of Otolaryngology, St. Louis, MO 63110)

Previously, we described a method for studying stimulus-induced release of substances from the *Xenopus* lateral line [Bledsoe *et al.*, J. Acoust. Soc. Am. Suppl. 1 65, S11(A) (1979)]. To date, 13 experi-

ments have been completed in which the release of endogenous amino acids was studied by comparing the effects of pulsatile water motion on the efflux of glutamate and aspartate from isolated skins with and without lateral-line stitches. For a given experiment paired stitch and nonstitch skins were subjected to two, 30-min periods of stimulation and two, 30-min periods of no stimulation in a rotated order-of-occurrence. In an overall comparison of mean amino acid levels for the first and second periods of the stimulated versus nonstimulated conditions, stimulation applied to the outer skin surface caused a significant increase in glutamate and aspartate efflux from stitch ( $p < 0.01$ ) and nonstitch ( $p < 0.01$ ) skins. The stimulated efflux of glutamate from stitch skins was significantly greater than the stimulated efflux from nonstitch skins ( $p < 0.01$ ). For aspartate efflux the between skin comparison was less clear cut with stimulation causing a significantly greater efflux from stitch skins than from nonstitch skins ( $p < 0.01$ ), only when comparing the mean aspartate levels for the first periods of the stimulated versus nonstimulated conditions. Although the source and mechanisms underlying the greater stimulated efflux from stitch skins remain to be determined, the results demonstrate differences between stitch and nonstitch skins under stimulus conditions which should induce release of the afferent transmitter substance(s). [Work supported by NIH and The Kresge Foundation.]

**X6. Effects of acute noise traumata on cochlear frequency analysis as observed in whole-nerve and single-cell recordings.** E. van Heusden and G. F. Smoorenburg (Institute for Perception TNO, Soesterberg, The Netherlands)

Tuning curves were obtained in cat by masking the whole-nerve action potential (AP), applying a forward-masking paradigm. The  $Q_{10}$  values of the tuning curves appeared to be about half the values found for threshold tuning curves of single nerve fibers. After inducement of a noise trauma by a 105 dB SPL wide-band noise that lasted for 30 min,  $Q_{10}$  values of the AP tuning curves became noticeably smaller. We carefully checked that this decrease of  $Q_{10}$  should be attributed to a loss in tuning selectivity and not to changes in the masking phenomenon itself. A decrease of  $Q_{10}$  values due to the noise trauma was found also for tuning of single cells in the antero-ventral cochlear nucleus. For both the whole-nerve and the single-cell responses, the decrease of  $Q_{10}$  is not simply an effect of stimulus level. The decrease is found for tuning curves just above pre- and post-trauma thresholds as well as for tuning curves at the same SPL before and after inducement of the trauma. [Research supported by the Netherlands Organization for the Advancement of Pure Research, ZWO.]

**X7. The acoustical inverse problem for the cochlea.** Man Mohan Sondhi (Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974)

A knowledge of the spatially varying mechanical properties of the basilar membrane (BM) allows one to compute acoustical properties of the cochlea, e.g., the transfer function from stapes to a point on the BM. However, to the best of our knowledge, direct measurements of BM properties have not been reported in the literature. Rather, these properties are "estimated" by trial and error so as to give a reasonable match to measurements of the above-mentioned transfer function. In this talk we will show how the BM stiffness  $K(x)$  may be determined directly from a time-domain measurement of the input impedance at the stapes (i.e., from a measurement of the pressure developed just inside the stapes in response to an impulse of stapes velocity).

**X8. A mode of peripheral auditory processing.** R. A. Houde (Center for Communications Research, 1531 St. Paul Street, Rochester, NY 14621), J. C. Bancroft (National Technical Institute for the Deaf,

Rochester, NY 14623), and C. W. Parkins (ENT Department, Upstate Medical Center, Syracuse, NY 13210)

A model of peripheral auditory processing was developed to guide the experimental design of a multichannel cochlear implant. The model specifies the neural processing necessary to represent observed PST histogram behavior of single eighth-nerve neurons and its relationship to the design of a multichannel vocoder processor. The model includes electrical characteristics of the cochlea. Simulation studies of stimulation characteristics are reported.

**X9. Modulation of cochlear nerve fiber activity by low frequencies.** K. Ingvarsson and P. Dallos (Auditory Physiology Laboratory, Northwestern University, Evanston, IL 60201)

A 100-Hz tone of relatively high intensity (40–80 dB SPL) has a modulating effect on the activity of auditory fibers when the cochlea is stimulated with this tone and a tone at the characteristic frequency (CF). This effect can be noted even when the low-frequency tone itself does not stimulate the fiber. The phase of the 100-Hz tone at which enhancement of activity driven by the CF-tone is seen to occur corresponds to the phase where the 100-Hz tone alone elicits activity. In fibers of CF greater than 2 kHz, the velocity component of the 100-Hz tone is not appreciable, compared to the CF velocity component. Thus the effect described lends support to the theory that the static position of the basilar membrane has a biasing effect on the inner hair cell excitation. [Supported by grants from the NINCDS.]

**X10. Two-tone suppression in the auditory nerve of the cat: Suppression thresholds and rate of growth.** Eric Javel (Human Communication Laboratories, Boys Town Institute, Omaha, NE 68131)

Our previous work has been directed at elucidating the behavior of the two-tone suppression phenomena when both tones are excitatory. Analyzing the phase-locked response of fibers possessing low characteristic frequency (CF), we have shown [E. Javel, *J. Acoust. Soc. Am. Suppl. 1* **65**, S83(A) (1979)] that suppression exists throughout an auditory nerve fiber's response area and that the ability of an excitatory tone to suppress the response to another excitatory tone is similar to a nonexcitatory tone's ability to suppress. For a given suppressor-tone intensity, suppression is maximal when the suppressor tone lies near fiber CF, and suppression magnitude is reduced on either side of CF. The contours relating suppression magnitude to frequency are similar in shape, but not in extent, to the isointensity response contours observed in determinations of response area. The "threshold" for suppression is systematically related to fiber sensitivity at the suppressor-tone frequency, but it is always greater or equal to the pure-tone threshold at that frequency. Threshold for suppression is not related to the threshold for the tone being suppressed. Once the threshold is exceeded, suppression for both excitatory and nonexcitatory tones grows with increasing suppressor-tone intensity in a manner consistent with that described by Sachs and Abbas [*J. Acoust. Soc. Am.* **60**, 1157–1163 (1976)]. The final slope of the curve relating suppression magnitude to suppressor-tone intensity is near unity for suppressor frequencies near fiber CF, but is usually greater than one for suppressors much lower in frequency than CF and less than one for suppressors higher in frequency than CF. Suppression does not saturate at high intensities, but in many instances its magnitude is obscured by the "notch" often seen in single-tone rate-intensity functions [Kiang *et al.*, *J. Acoust. Soc. Am.* **46**, 106(A) (1969)], i.e., the "notch" cannot be suppressed. [Work supported by NIH.]

**X11. AP responses in forward-masking paradigms and their relationship to responses of auditory-nerve fibers.** Paul J. Abbas and Michael P. Gorga (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

The amplitude of  $N_1$  peaks of whole-nerve AP was measured in cats using a tone-burst probe with tonal maskers in a forward-masking paradigm. Experiments examined the effects of masker level, frequency, and duration ( $T_m$ ), as well as interstimulus interval ( $\Delta t$ ). Results are consistent with the interpretation that amplitude of  $N_1$  reflects activity of a limited group of fibers with CF near the probe frequency. For single fibers, decrement in discharge rate (relative to an unmasked condition) is dependent upon rate to the masker [Smith, J. Neurophysiol. 40, 1088–1112 (1977)]. For the AP, decrement in probe-elicited amplitude should reflect the rate to the masker in those fibers excited by the probe. Thus, measurements of  $N_1$  decrement *versus* masker frequency and level are similar to single-fiber rate-*versus*-level functions. Since decrement in  $N_1$  reflects amount of adaptation, increasing  $T_m$  decreases probe response.  $N_1$  amplitude as a function of  $T_m$  thus resembles PST histograms of nerve fibers. Finally, plots of  $N_1$  amplitude as a function of  $\Delta t$  are interpreted as a measure of recovery from adaptation.

**X12. AP tuning curves and anatomical correlates from acoustically traumatized cats.** N. T. Shepard and P. J. Abbas (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Compound action potential tuning curves, using a forward-masking paradigm, were developed on both a control group and a group of acoustically traumatized cats. Differences observed between the two populations included a decrease in the sharpness of the tip, in the sensitivity of the tip, and/or in the sensitivity of the tail region. Phase contrast light microscopy was performed on all exposed ears using a celloidin embedding technique with horizontal sectioning. Whenever an abnormality in an AP tuning curve was seen, histological evidence of damage to the organ of Corti in an appropriate region corresponding to the signal frequency was observed. However, several cases of damage to the cochlea were observed with normal tuning curves. Wherever the tip region of the tuning curve was elevated, evidence of damage to all three rows of OHC and to the IHC was seen. Several other observations relating to the extent of damage to the OHC and IHC independently were found to be similar to results from single-fiber recordings in acoustically traumatized cats [M. C. Liberman and N. Y. Kiang, *Acta Otolaryngol. Suppl* (STOCKH) (358), 1–63 (1978)].

**X13. High-frequency sensitivity of the frog basilar papilla confirmed by dye injection.** E. L. Leverenz and E. R. Lewis (Biophysics Group, University of California, Berkeley, CA 94720)

High-frequency sensitivity (cf. in the range 1000 to 2000 Hz) has been confirmed in the basilar papilla of the bullfrog by recording the responses of single eighth-nerve axons, then injecting those axons with the fluorescent dye Lucifer Yellow, and thereby tracing them to their origins at the periphery. Dye-filled glass micropipettes were used to penetrate primary afferent axons in the posterior branch of the eighth cranial nerve medial to its emergence from the intact otic capsule with intact circulation. All axons that we have traced to the basilar papilla have been of the high-frequency type [Feng *et al.*, *J. Comp. Physiol.* 100, 221–229 (1975)], while all units that we have traced to the amphibian papilla have been of the low- or mid-frequency type. So far, all primary afferent axons traced to the basilar papilla have been unbranched and have innervated one or perhaps two hair cells. Our direct observations of high-frequency sensitive fibers emanating from the basilar papilla confirm the conclusions drawn previously by Frishkopf, Capranica, and Goldstein [Proc. IEEE 56, 969–980 (1968)]; Fend *et al.* (op. cit.); and others.

**X14. Some new causes of acoustic impedance change during acoustic stimulation in man.** V. R. Singh (National Physical Laboratory, Hill Side Road, New Delhi 110 012, India)

Investigation of the actual nature and causes of the acoustic impedance change during acoustic stimulation in the ear has been an active area in the recent past. It is believed that the impedance change is only due to the middle ear muscle activity, but according to the present investigation, this is not true. The acoustic impedance has been studied with and without ipsilateral sound stimulus by using an electroacoustic bridge technique in living human ears, temporal bones, and mechanical models of the ear. The change in the acoustic impedance is also found in denervated cadaver bones and mechanical ears. This clearly shows that the muscle activity is not the only cause, but some new causes like physiological characteristics of the eardrum, anatomical structures of the ear, local induction phenomenon in the ear, and compliance of the eustachian tube are also responsible for the acoustic impedance change during the sound stimulation. The present study is useful in investigating the actual nature and causes of infections and diseases of the ear. [Work supported by Canadian Commonwealth Scholarship and Fellowship Committee.]

WEDNESDAY AFTERNOON, 28 NOVEMBER 1979 GRAND BALLROOM III, 2:00 TO 4:35 P.M.

## Session Y. Speech Communication IV: Perception of Suprasegmental Cues

Cecil H. Coker, Chairman

*Bell Laboratories, Murray Hill, New Jersey 07974*

Chairman's Introduction—2:00

### *Contributed Papers*

2:05

**Y1. Psychoacoustic model of categorical perception for speech and nonspeech continua.** Blas Espinoza-Varas (Nav. Sub. Medical Research Laboratory, Auditory Department, Submarine Base, Groton, CT 06340)

The model assumes: (1) that categorization of a stimulus continuum  $x_1, x_{i+1}, \dots, x_n$  is made according to whether the stimuli fall above or

below a decision "boundary,"  $x_B$ ,  $i < B < n$ ; and (2) that discrimination between any two stimulus values,  $x_p, x_r$ , of the continuum is based on the physical parameter  $\phi$ , where  $\phi = |x_p - x_r|/|x_r - x_B|$ ;  $x_p$  is the member of the stimulus pair that is proximal to the boundary, and  $x_r$  the one that is remote from it. That is, discrimination is based on the distance between the pair of stimuli, proportional to the distance between the boundary and the remote stimulus of the pair. For both speech and nonspeech continua it is found that, given a distance

$|x_p - x_r|$ , the parameter  $\phi$  reaches a sharp maximum for stimuli near the boundary,  $x_b$ , and decreases monotonically as the distance between the remote stimulus of the pair and the boundary increases. The model was used to predict discrimination data for VOT [Wood, J. Acoust. Soc. Am. 60, 1381–1389 (1976); Elman, J. Acoust. Soc. Am. 65, 190–207 (1979)], onset of  $F_2$  transition and duration of stop closure [Elman, *ibid.*], and noise-buzz sequences [Miller *et al.*, J. Acoust. Soc. Am. 60, 410–417 (1976)]. For most of these data, discriminability appeared to be an increasing function of  $\phi$ . A corollary of the model is that enhanced discriminability around the boundary is a consequence of the listener adopting such boundary, rather than vice versa. [Supported by NRC Postdoctoral Fellowship to the author.]

2:20

**Y2. Perceptual equivalence of cues for a phonetic trading relation: Primacy of phonetic over psychoacoustic effects.** Catherine T. Best (Haskins Laboratories, New Haven, CT), Barbara Morrongiello, and Rick Robson (University of Massachusetts, Amherst, MA 01003)

Recently a phonetic trading relation was found for two different cues to the identification of [split] versus [slit], namely, silent closure duration versus initial formant transition differences for the [plit]–[lit] contrast. Within limits, varying either cue produces the phonetic contrast; an extra amount of one compensates for lack of the other. [Erickson, Fitch, Halwes, and Liberman, J. Acoust. Soc. Am. Suppl. 1 61, S46(A) (1977)]. Subsequently, a three-way discrimination test was devised to more clearly show that the phonetic effects of the independent cue manipulations are perceptually equivalent [Fitch, Halwes, Erickson, and Liberman, Haskins Laboratories: Status Report (1979)]. The researchers hypothesized that the trading relations and perceptual equivalence phenomena reflect perceptual specialization that takes account of the common origin of cues in single articulatory gestures. But the phonetic generality of the paired phenomena needs assessment; therefore, we used their procedures to test the simpler [stay]–[say] contrast. Both phenomena were replicated and virtually identical in magnitude to those reported earlier. To test the phonetic specificity of the phenomena, we ran a perceptual equivalence discrimination test with nonspeech sine-wave analogs of the stimuli, using naive subjects. Results suggest that, in perceiving stimuli, *speech* sharply increases the effectiveness and shifts the absolute value, of a weak psychoacoustic boundary. [Work supported by NICHD.]

2:35

**Y3. Difference limens for fundamental frequency contours in sentence level stimuli.** M. S. Harris and A. M. S. Quinn (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

Two experiments were performed to measure the difference limen (DL) for fundamental frequency using sentence level speech stimuli. In experiment I a male reader recorded five sentences. They were analyzed and synthesized using a linear prediction vocoding system. Part of each sentence was varied in 5-Hz steps above and below the original contour. Sentences (original contour and one of the changed contours) were presented to listeners using the method of constant stimuli. DL's ranged from less than 5 to 15 Hz depending on the sentence. There was no significant difference between DL's as a function of direction of change (up or down from original). Experiment II was an attempt to determine if the variability in DL due to sentences was a function of differences in placement of the frequency change. The changes in fundamental frequency began and ended at voiceless portions in all sentences. The median DL for the speaker also used in experiment I was 12 Hz. No significant differences as a function of sentences were found. For another speaker the DL's were over 15 Hz.

2:50

**Y4. Determination of temporal discriminations by an adjustment procedure.** T. D. Clack (Department of Otorhinolaryngology, University of Michigan, Ann Arbor, MI 48109) and P. J. Benson (Phonetics Laboratory, University of Michigan, Ann Arbor, MI 48109)

Identification of initial stop consonants along the voicing continuum exhibits sharp category boundaries as demonstrated with experimental manipulations of the voice onset times of natural and synthetic speech. The existence of discrimination thresholds for temporal onset differences has been suggested as a simple explanation of this phenomenon. Evidence of such a threshold for simple nonspeech sounds has been obtained using the ABX method, but with considerable inconsistencies among listeners [D. Pisoni, J. Acoust. Soc. Am. 61, 1352–1361 (1977); P. J. Benson and T. D. Clack, J. Acoust. Soc. Am. Suppl. 1 65, 58(A) (1979)]. This variability might be primarily methodological in origin and consequently a different psychophysical procedure has been developed. Discrimination thresholds were determined by an adjustment procedure similar to Bekesy audiometry. The results for the same subjects and listening conditions are evaluated relative to our previous ABX results.

3:05

**Y5. Allophonic, stress, and prosodic cues for word perceptions.** Lloyd H. Nakatani and Kathleen D. O'Connor–Dukes (Bell Laboratories, Murray Hill, NJ 07974)

Speech is apparently rich in phonetic parsing cues which enable listeners to recover the words of an utterance. Listeners easily distinguished between phrases such as "stronger ranger"/"strong arranger" and "maiden forced"/"maid enforced." No contextual cues were available, so listeners had to use phonetic cues to distinguish the phrases. By acoustic analyses and careful listening, phonetic differences between the phrase pairs were identified; these differences constituted the possible parsing cues. Of these, allophonic and other segmental variations were the most salient parsing cues; stress pattern variations were also salient, but prosodic variations in pitch and rhythm were not very salient.

3:20

**Y6. The role of morphology in a word boundary decision task.** Kathleen D. O'Connor–Dukes and Lloyd H. Nakatani (Bell Laboratories, Murray Hill, NJ 07974)

We investigated how listeners use their knowledge of English morphology to divide nonsense phrases into "words." We looked at two types of syllables which occur as affixes in English. One type occurs as both morphological prefixes and suffixes, such as /di/ in "trema demarneted"/"tremady marneted." Another type occurs only as prefixes, such as /pro/ in "cler proventched" but not \*"clerpro ventched"; or only as suffixes, such as /bl/ in "afeemable talse" but not \*"afeema bultalse." We found that morphological considerations affected listeners' judgments of boundary location when phonetic differences between the phrases were weak or absent. But when there were strong phonetic differences between the phrases, these differences dominated the perception of word boundaries, overriding the influence of morphological effects.

3:35

**Y7. Syllable durations and phonological groups.** I. D. Conday (Department of Linguistics, University of Hawaii, Honolulu, HI 96822)

The phonological organization of Fijian can be accounted for without positing the *word* as a phonological unit. Instead, the units *syllable* and *measure* appear adequate to account for much of the phonol-

ogy [A. J. Schutz, Univ. Hawaii Working Papers in Linguistics 8.2, 75–100 (1976)]. In this system the phonological measure is a group of syllables; the boundaries of the measure need not be coterminous with the boundaries of morphemes. The more syllables in a measure, the shorter each syllable is. This result is comparable to findings for English words [I. Lehiste, in Lass (ed.), 225–229 (1976)]. Examples in support of this analysis are of the type given below: an underlyingly long vowel in a morpheme such as *sā* remains long when it is in a measure by itself. When *sā* joins another syllable in making a disyllabic measure, the *sā* is shortened, whether or not the following syllable is a monosyllabic morpheme or part of a polysyllabic morpheme. The shortened *sā* is never as short as an underlyingly short *sā*, which would be a different morpheme.

3:50

**Y8. Effect of talker sex on sentence discrimination scores.** S. E. Gerber and D. A. Yrchik (Speech and Hearing Center, University of California, Santa Barbara, CA 93106)

The effects of talkers' sex on sentence discrimination scores of normal hearing listeners were examined. Two talkers each recorded ten synthetic sentences and a competing story. From this, four talker–masker combinations were prepared: male talker, male masker; male talker, female masker; female talker, female masker; female talker, male masker. Each combination was presented with the talker and masker in the same ear and in one ear only. Three different message-to-competition ratios (+10, 0, –10 dB) were used to present the four talker–masker combinations to 16 listeners. At each level, the talkers' sentence stimuli remained fixed at 60 dB SPL while the competing story was varied from 50 to 70 dB. Statistical analyses of the results indicated that significant differences in discrimination were produced by the message-to-competition ratios, the talker–masker combinations, and the interaction between them. Implications of these differences are discussed in the context of these results.

4:05

**Y9. Children's detection of mispronounced words in fluent speech.** Maria Kulick, Beatrix Wiedmer, and Patricia Kuhl (Department of Speech and Hearing Sciences, Child Development and Mental Retardation Center, University of Washington, Seattle, WA 98195)

Children's ability to recognize mispronounced words in fluent speech was measured using an error-detection task in which the children rang a bell when they heard a mispronounced word in a story. Mispronunciations of the same word occurred in contexts in which they were semantically predictable and semantically unpredictable. The mispronunciations occurred word-initially on stop consonants and consisted of changes in voicing or place of articulation. Data were analyzed for the effects of semantic context, feature change, age, and sex on the percentage of errors detected and on the latency of response. Results indicated that four- to six-year-olds were very accurate (nonsignificant age differences) at detecting mispronunciations ( $\geq 78\%$ ). Detection of mispronunciations in semantically predictable contexts were significantly better (and latencies were significantly shorter) than in unpredictable contexts, for all age groups. There were no significant differences in the overall detection of mispronunciations involving a change in voicing versus place; however, significant differences for direction of change within feature occurred for both voicing and place. Females were significantly more accurate than males at detecting mispronunciations.

4:20

**Y10. Adaptation of speech by nonspeech: Evidence for complex acoustic cue detectors.** Arthur G. Samuel (Bell Laboratories, 600 Mountain Avenue, Murray Hill, NJ 07974) and Elissa N. Newport (Department of Psychology, University of Illinois, Champaign, IL 61820)

Three selective adaptation experiments were run, using nonspeech stimuli (music and noise) to adapt speech continua ([ba]–[wa] and [cha]–[sha]). The adaptors caused significant phoneme boundary shifts on the speech continua only when they matched in periodicity: Music stimuli adapted [ba]–[wa], whereas noise stimuli adapted [cha]–[sha]. However, such effects occurred even when the adaptors and test continua did *not* match in other simple acoustic cues (rise time or consonant duration). Spectral overlap of adaptors and test items was also found to be unnecessary for adaptation. The data support the existence of auditory processors sensitive to complex acoustic cues, as well as units that respond to more abstract properties. The latter are probably at a level previously thought to be phonetic. Asymmetrical adaptation was observed, arguing against an opponent-process arrangement of these units. A two-level acoustic model of the speech perception process is offered to account for the data.

WEDNESDAY EVENING, 28 NOVEMBER 1979

EVENING

6:30 Social Hour. Grand Ballroom III

7:30 Banquet. Awards. Grand Ballroom II

Presentation of the Wallace Clement Sabine Medal to Cyril M. Harris

Presentation of the Silver Medal in Physical Acoustics to Herbert J. McSkimin

8:30 Entertainment. International Ballroom Dance Company of Brigham Young University. Grand Ballroom II



**Session Z. Architectural Acoustics III: Miscellaneous**

M. David Egan, Chairman

*College of Architecture, Clemson University, Clemson, South Carolina 29631***Chairman's Introduction — 9:00****Contributed Papers****9:05**

**Z1. Direct Measurement of Wall Reflections in an Anechoic Chamber.** Gabriel Weinreich and Eric B. Arnold (Randall Laboratory of Physics, University of Michigan, Ann Arbor, MI 48109)

The system developed for measuring the radiation fields of a violin [G. Weinreich and E. B. Arnold, *J. Acoust. Soc. Am. Suppl.* 1 65, S72(A) (1979)] allows a direct measurement of the angular distribution of amplitude and phase of the wave reflected from the wall of an enclosure. By means of a special boom system, two microphones sweep out two concentric spheres surrounding the source of sound. The data are recorded in computer-readable form and then analyzed. By expanding in spherical harmonics and corresponding Hankel functions, the outgoing (transmitted) and incoming (reflected) components are separately determined. Data will be shown for the anechoic chamber of the University of Michigan Phonetics Laboratory [G. W. Peterson, G. A. Hellwarth, and H. K. Dunn, *J. Audio Eng. Soc.* 15, 67–72 (1967)]. [Work supported by NSF.]

**9:20**

**Z2. Acoustical evaluation and control of normal room modes.** Oscar J. Bonello (Department of Research, Solidyne SRL., Tres de Febrero 3254, (1429) Buenos Aires, Argentina)

Analysis of a new criterion for the correct distribution of normal room modes is presented. This criterion replaces unsuccessful earlier ones. It is based on the resonance modes density curve related to one-third octaves. A computer program, easy to perform, allows to obtain the best dimensional ratio of the room, to avoid sound coloration. This method was duly verified, applying it to the calculation of the dimensions of a high number of existing studios considered of excellent quality and of others with evidence of coloring. The correspondence of theoretical and practical conclusions was very good. As second step, it will describe a special device for acoustical absorption. This resonant absorber is frequency selective with high absorption in the low-frequency end, remaining low and constant beyond 2000 Hz. By means of a computer program, the spatial distribution of the isolated room modes can be plotted. It allows to place the absorption devices in the exact position to obtain maximum control of each mode.

**9:35**

**Z3. An experimental and computer study of the spatial distributions of  $p^2$  in a reverberant enclosure at low frequencies.** J. B. Ochs (Department of Mechanical Engineering and Mechanics, Lehigh University, Bethlehem, PA 18015) and J. Tichy (The Pennsylvania State University, University Park, PA 16801)

In a diffuse sound field the spatial distributions of  $p^2$  are exponential. Because in an enclosure the diffusion of the sound field is not perfect, it is necessary for some applications, such as the sound power measurements, to know how much the statistical distributions deviate from exponential, particularly at frequencies lower than the

critical frequency  $f_c = 2000 (T/V)^{1/2}$ . Therefore, both computer and experimental studies were conducted from below the first mode in an enclosure up to and above the critical frequency in 1-Hz frequency increments with  $p^2$  calculated and measured in 120 spatial positions. Both probability density functions and cumulative distribution functions as well as the deviation from the exponential distributions were obtained. Computer-generated second order statistics have been investigated to determine the effect of a finite number of points used to generate the distributions. Based on the frequency dependence of the deviation of the actual distributions from the exponential distributions, it can be concluded that the reverberation room is usable from a lower frequency than  $f_c$ .

**9:50**

**Z4. Sound decays in a reverberation room.** A. C. C. Warnock (Division of Building Research, National Research Council of Canada, Ottawa, Ontario, Canada K1A 0R6)

The decay of the sound field in the reverberation room at the NRC has been examined using a computer-controlled real-time analyzer. This system provides the means of studying in detail the spatial variations of reverberation times and decay linearity. Many decays were recorded for a number of diffuser systems and microphone positions including corner microphones. The results obtained are compared with the requirements on spatial variation and linearity in ASTM method of test C423. The requirements are not easily met.

**10:05**

**Z5. Study of the sound transmission class system for rating building partitions: Another view.** Ludwig W. Sepmeyer (Consulting Engineer, 1862 Comstock Avenue, Los Angeles, CA 90025)

In an earlier paper titled "Subjective Study of the Sound Transmission Class System for Rating Building Partitions," D. M. Clark concluded that his results showed that the ASTM E413 method for determining a rating from measured transmission-loss data yields conservative values even when there are large dips in the TL characteristic. Clark's experimental procedure had the subjects match the annoyance produced by a degraded STC shaped sound channel by increasing the level transmitted by a pure STC contour shaped channel. By this method, instead of having a fixed reference annoyance, the reference first established in the pure STC channel is increased in proportion to frequency, bandwidth, and depth of the degradation introduced into the so-called TL channel. If instead the reference annoyance had been kept fixed and the degraded TL channel had been readjusted to match the chosen criterion value of annoyance, quite different results would have been obtained. By using speech articulation index computations, it is shown that instead of being conservative, the rating procedure can overrate TL characteristics with high-frequency octave band dips as much as five points. Instead of abandoning the 8-dB limitation completely, it is proposed that a limitation of 16 dB for the sum of any three adjacent bands be substituted.

10:20

**Z6. Sound transmission loss of windows.** J. D. Quirt (Division of Building Research, National Research Council of Canada, Ottawa, Canada K1A 0R6)

An extensive series of measurements has been made to obtain a systematic evaluation of the effects of glass thickness and interpane spacing on the sound transmission loss of windows. These included 93 laboratory measurements conforming to ASTM standard E90 and field measurements on a subset of 18 of these window types. The samples studied included glass thicknesses from 3 to 6 mm and interpane separations from 6 to 150 mm. In addition, the effects of absorptive treatment, cavity dividers, and nonparallel glass surfaces were investigated. The data contradict one widely held belief about window design.

10:35

**Z7. St. George's Theatre, London.** Jeff Charles (Bickerdike Allen Partners, 16 New End, London NW3 1JA, England)

St. George's Church was consecrated in 1867. The designer was George Truefit, who, it is said, modeled the Church on the 5th century Crusader church of St. George in Salonica. The church survived two world wars, although some damage was caused by flying bombs in 1944. After the war the church became derelict and empty, and the St. George's Elizabethan Theatre Limited Organization produced plans for the conversion of the church to a theatre. After purchasing the church in 1973, the theatre opened to the public on St. George's Day for a 6-month Shakespearean festival. Regrettably the acoustics left very much to be desired and the first season was marred by a considerable volume of complaints from the audience after performances and critics' headline comments such as "The Play's the Thing—but I could not hear it." The paper discusses the remedial acoustic measures that have been undertaken to make the theatre acceptable.

10:50

**Z8. The successful auditorium: Some acoustics and a lot of psychology.** Philip Dickinson (Bickerdike Allen Partners, 16 New End, London NW3 1JA, England and 442 North Main Street, Bountiful, UT 84010)

Remedial measures to a large, well-known concert hall produced acoustics that, the acoustical consultants believe, are as good as any in the world today. Performers, concert goers, and critics were unanimous in their expressions of delight during the first concert season. There were high hopes of it becoming one of the world's great auditoria. But, all is not well and opinions are changing even though the acoustic characteristics remain unchanged. The paper discusses the other factors involved in producing a successful auditorium and suggests reasons for the demise of this hall and for the popularity of some others in which the acoustics, to be polite, can only be described as "interesting."

11:05

**Z9. Consulting on more than a building a day.** David E. Drommond (The Church of Jesus Christ of Latter-day Saints, 50 East North Temple Street, Salt Lake City, UT 84150)

Acoustical Engineers are normally able to spend considerable time analyzing a room in order to provide a satisfactory acoustical space; however, such time cannot be spent when faced with the large number of buildings that the Church erects each year. Designs must be "mass produced," yet each building should also receive individual attention. Standard acoustical practices have been modified to meet these peculiar needs. This paper summarizes the design, review, construction, and inspection phases of a typical meetinghouse designed for the high volume mode.

11:20

**Z10. A case study of acoustical performance of prefabricated music practice rooms.** Charles R. Boner (Boner and Associates, Austin, TX 78767)

The basis for procurement of 115 rooms for the new music building for the University of Texas at Austin College of Fine Arts/Performing Arts Center, now under construction, consisted of noise control testing and interior noise level testing of rooms in the plants of four manufacturers. The results of the testing were of invaluable aid in preparing specifications for procurement. The rooms were installed in temporary facilities while the new building is under construction and will all be relocated to the new building following its completion.

THURSDAY MORNING, 29 NOVEMBER 1979

BONNEVILLE ROOM 4, 9:00 TO 11:50 A.M.

## Session AA. Engineering Acoustics V: Digital Audio Processing for Quality Sound Reproduction (Session dedicated to Benjamin B. Bauer)

Daniel Queen, Chairman

*Daniel Queen Associates, 5524 West Gladys Avenue, Chicago, Illinois 60644*

Chairman's Introduction—9:00

### *Invited Papers*

9:05

**AA1. Benjamin Bauer's contribution to audio and acoustics.** Louis A. Abbagnaro (CBS Technology Center, Stamford, CT 06905)

Benjamin Bauer worked actively in Audio and Acoustics for over 40 years. He made contributions in the areas of engineering acoustics, psychoacoustics, and underwater acoustics. This paper will describe selected major topics in those areas which are still of interest today.

9:35

**AA2. Digital recording of audio for commercial release.** T. G. Stockham, J. I. Bloomenthal, R. B. Ingebretsen, and B. C. Rothoar (Soundstream Incorporated, 34 South 600 East, Salt Lake City, UT 84102)

Several hundred commercial recordings, released or scheduled for release, have already been made by a handful of companies worldwide. This paper will discuss some technical aspects of those recordings made by the author's group and will present a high quality demonstration of a few selected samples.



10:05

**AA3. Amplitude and phase accuracy of broadcasting microphones.** Louis A. Abbagnaro (CBS Technology Center, Stamford, CT 06905)

Phase and amplitude accuracy of microphones can influence the quality of recorded programs. These effects have become more pronounced as multichannel recording has become an accepted format, and they can degrade both the frequency response and spatial accuracy of the information. Techniques to overcome such problems, include special microphone designs such as multielement microphones, "sound field" microphones, and the use of artificial heads. Strengths and weaknesses of several approaches are reviewed.

10:35

**AA4. Digital technology and acoustical accuracy in domestic music reproduction.** Robert Berkovitz (Teledyne Acoustic Research, Norwood, MA 02062)

Accuracy of spectral magnitude measurement is a generally accepted criterion of sound reproduction ability, but there is little correspondence between the conditions under which usual loudspeaker measurements are made and the realities of loudspeaker usage. It is also unclear whether specific programs are intended for *field* reproduction, in which the listening environment is cancelled, or *source* reproduction, from which the recording environment is absent. The achievement of either objective depends upon (1) the use of loudspeakers which meet certain directivity criteria and (2) new transmission systems which may be suitable applications of digital encoding. Loudspeaker measurements which employ digital signal processing may aid the design of such loudspeakers and transmission systems by allowing (1) acquisition of binaural data in actual room settings, (2) observation of the response of linear basilar membrane models to real sources, (3) extraction of useful interaural cross-correlation data, and (4) separation of room effects from those of direct source radiation. Examples of such measurements are shown.

11:05

**AA5. Direct digital conversion in acoustic transducers.** J. L. Flanagan (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

We offer some views, some computer analyses, and some preliminary experimental measurements on acoustic transducers designed to operate directly with digital signals. In particular, we describe a condenser receiving element and acoustic filter system that functions directly as a digital-to-analog converter (DAC). The transducer accepts the binary digital signal of PCM-encoded speech and produces the decoded analog acoustic output. We also suggest a design for a transducer that performs the reverse operation, that is, analog-to-digital conversion (ADC). Initial results are very tentative; many problems remain. But the techniques hint at routes to digital communication that can obviate, at least to some extent, the expense and complexity of conventional ADC and DAC circuitry.

## Contributed Papers

11:35

**AA6. The tempered Fourier transform.** Dale T. Teaney, Victor L. Moruzzi, and Frederick C. Mintzer (IBM Watson Research Center, Yorktown Heights, NY 10598)

$\frac{3}{2} = 2^{7.01955/12} \cong 2^{7/12}$ . The set of harmonics ( $l = 1, 2, 3$ ),  $1/K_l = (\frac{3}{8}, \frac{1}{4}, \frac{1}{8}) \cong (2^{8/12}, 2^{7/12}, 2)$ , when scaled by  $2^{3m/12}$ ,  $m = 1, 2, 3, 4$ , constitutes a logarithmically uniform set,  $f_{lm}$ , of 12 frequency classes per octave (8 va). The tempered Fourier transform samples a record  $x(t)$  into four independent parallel sequences  $x_m(i)$  at sampling rates  $2^{m/4}$ . In each 8 va,  $n$ , of each sequence,  $m$ , three Fourier coefficients,

$$X_m(i) = \sum_{j=0}^{N-1} x_m(j-i)W(i/NK_l)\exp - j(2\pi i/K_l)$$

are computed, where  $W(i/NK_l)$  is a window  $N$  cycles long, and

$$x_m^n(j) = \sum h_k x_m^{n+1}(2j-k)$$

is the data sequence in the  $n$ th 8 va after it has been low-pass filtered and decimated from the  $(n+1)$  8 va above by a half-band filter with impulse response  $h$ . With a suitable value of  $N$ , the array  $X_{lm}^n$  is statistically and dynamically equivalent to a  $\frac{1}{12}$ -8 va spectral analysis of  $x$ . Examples and variants of the method will be discussed.

**Session BB. Musical Acoustics IV: The Singing Voice**

Harry Hollien, Chairman

*University of Florida, ASB Room 50, Gainesville, Florida 32611*

**Chairman's Introduction—9:00**

***Invited Papers***

**9:05**

**BB1. An integration of current research on the singing voice.** E. Allen (Institute of the Crippled and Disabled, New York, NY 10010)

The extremely specialized voice use of singers has been the subject of a small but significant body of research during the last few years. Since this innovative work has such a short history, the integration of investigations of the voice subsystems and the incorporation of physiologic, acoustic, and perceptual data had to await the collection of sufficient material to be effective. Such integrations are the focus of this presentation. The ramifications of Sundberg's conclusions on the shape of the vocal tract during singing, Shipp's larynx height data, Hixon's chest wall positioning data, Baker's model, and Large's register work will be included, among others. In addition, the continuing challenges of the aesthetic/research interface and the need for more refined instrumentation will be addressed.

**9:25**

**BB2. Abstract withdrawn.**

**9:45**

**BB3. Vocal jitter in singers voice.** T. Murray (VA Hospital, San Diego, CA 92161)

Cycle-to-cycle variations in the period of the vocal fold waveform, known as jitter or frequency perturbation, are a natural quasiperiodic property of the glottal sound source. Excessive jitter in the voice signal represents an undesirable vocal parameter which the singer strives to minimize. This study examined the degree of frequency perturbation in female singers' voices during the production of sung and spoken vowels. Four female singers produced the vowel /a/ in one spoken and four sung conditions. Magnitude perturbation measures were obtained for a steady state portion of each sample. The samples were also submitted to a panel of listeners who were asked to rate the degree of roughness in each vowel. The perceptual results from those judges who performed reliably were compared to the measures of vocal jitter. The results indicate that singers' spoken samples contain less jitter than their sung vowels. The perceptual ratings indicated that listeners were not able to judge the group in relation to their perturbation measures.

**10:05**

**BB4. Elements of quality variation: voice modes and singing.** R. H. Colton and J. Estill (Department of Otolaryngology and Communication Sciences, Upstate Medical Center, Syracuse, NY 13210)

In this presentation, the results of several experiments on voice quality will be reviewed. In all these studies, the subjects produced four perceptually distinct voice qualities on each of several fundamental frequencies throughout the whole phonatory range. In one experiment which tested the perceptual distinctiveness of each quality, observers were presented stimuli from which all but spectral cues had been eliminated. Observers were able to recognize the four qualities even at extremes of fundamental frequency. Analyzed acoustically, the four qualities are distinguished by frequency bandwidth, location of resonant peaks, spectral envelope, and dynamic range. Physiologically, each quality is characterized by specific behaviors in the larynx and within the vocal tract. In one, the vocal tract is very large, and the vocal folds appear to approximate with very little mass and with a very short closed phase. In another, the pharynx is small, with rapid opening and closing of the vocal folds. In yet another quality, the vocal tract is very small, with a peculiar configuration of the supraglottal structures. These and other observations are indicative of the differences among these qualities. Each quality represents a particular set of the vocal mechanism or mode of voice use. From the results of more recent experiments, we have developed the concept of voice modes to be unique sets of perceptual, acoustic, and physiological elements. These elements in the perceptual domain, for instance, are very much like distinctive features, except that they are probably not binary phenomena. Depending on the type of combination and the degree to which they are used, these "voice quality

features" can be used to specify the voice quality. This concept will be discussed in this paper. Voice modes and the concept of distinct physiological, acoustical, and perceptual features should be useful and meaningful to any singer. With these concepts, he or she can quickly learn the proper antecedent physiological conditions necessary for a specified vocal quality. Once mastered, he or she can vary elements of the voice mode set to produce subtle or gross differences of vocal quality that can result in greater variety in performance.

10:25

**BB5. A physiological interpretation of vocal registers.** I. R. Titze (Department of Speech Pathology and Audiology, University of Iowa, Iowa City, IA 52242)

Vocal registers have been described acoustically and perceptually, but little is yet known about register control at the muscular level. An attempt is made here to describe vocal registers as stable or transitional regions in a four-dimensional space consisting of cricothyroid, thyroarytenoid, interarytenoid, and pulmonary stress. The chest register is shown to be the central and most stable region where conditions for oscillation are optimal. Falsetto and glottal stop are other physiologically stable and robust regions, but conditions for oscillation are less favorable. Vocal fry and creaky phonation tend to manifest themselves as transitional states between stop, chest, and falsetto. Glottographic, electromyographic, and computer simulation data are combined to show the relationships between acoustic, configurational, and muscular characteristics of vocal registers. [Work supported by NINCDS.]

10:45

**BB6. Studies of the Garcian model for vocal registration.** J. W. Large (School of Music, North Texas State University, Denton, TX 76203)

This paper reports a series of studies in which a model for vocal registration central to the well-documented vocal method of Manuel García (1905–1906) was utilized. García was the teacher of Jenny Lind, Mathilde Marchesi, and Julius Stockhausen and the inventor of the laryngoscope. Equalized and nonequalized paired tones (same fundamental frequency, sound level—A-weighting, and vowel—/a/) in the areas of register overlap in both male and female premier singers were recorded (1) on a high-fidelity tape recorder for perceptual evaluation and Sonagraphic analysis, (2) by a pneumotachographic system for measurement of airflow rates, and (3) with photography for analysis of laryngeal movements. The results of these investigations suggest the existence of physical correlates for singers' subjective sensations of different registers. These sensations coupled with the percepts of expert vocal pedagogues are responsible for the continued use of Garcian terms such as "chest" register, "falsetto" register, and "head" register.

11:05

**BB7. Elements of frequency and amplitude modulation in the trained and pathologic voice.** T. Shipp and K. Izdebski (VA Medical Center and the University of California, San Francisco, CA 94121)

Rhythmic changes in the frequency and/or intensity of the vocal signal during sustained phonation is characteristic of trained singers and individuals with some neurological dysfunction. The questions that are explored in this presentation are: (1) For the two groups, what aspect of the acoustic signal is being modulated? (2) Is the mean repetition rate similar for the two groups? (3) How regular are the cyclic variations in the signal between the two groups? and (4) What physiologic assumptions can be made about the origins of these phenomena? Acoustic analysis of sustained vocal output by ten professional opera singers producing frequencies throughout their vocal range and the sustained vocal phonation by selected patients from a population of spastic dysphonic individuals indicates two differences between the groups: Singers primarily vary the frequency of phonation  $\pm 0.5$  semitone for vibrato, whereas the patients primarily varied signal amplitude. The mean rate at which the variations took place were the same for the two groups. The variability of cycle-to-cycle measures was extremely small in the professional singers and quite large for the patients. The vibrato rates obtained were substantially below those given as normal physiologic tremor rate. Similarity in the rates between the two groups suggests that the physiologic mechanism for vocal tremor and vibrato may be the same with the singers demonstrating an ability to inhibit the pulses in all muscle groups except for the superior laryngeal nerve innervating the cricothyroid muscle. Patients and, perhaps, less skilled singers demonstrate periodic muscle stimulation in sites throughout the vocal tract including the respiratory system.

11:25

**BB8. Objective measurements of the singer's voice as a "damage risk" indication.** R. F. Coleman (The Bill Wilkerson Hearing and Speech Center, Vanderbilt University, Nashville, TN 37212)

Professional singers, particularly beginning performers, seldom have any objective means to determine safe performance limits of their voices, and so must rely on trial and error experience to know

what singing ranges are feasible for the individual. This almost invariably leads to vocal difficulty, ranging from simple laryngitis to chronic, profession threatening vocal pathology. This presentation will discuss normative data on the singer's voice, particularly with respect to the physiological and acoustic limits of the instrument measured in a " $F_0$ -SPL Profile." Examples of professional performers' voice profiles will be used to illustrate the usefulness of the technique in estimating a type of damage risk criteria for voices.

11:45

**BB9. Studies of pulmonary function and the singer.** W. J. Gould (Lenox Hill Hospital, New York, NY 10021)

The relationship of the efficiency of pulmonary function to the singing voice was studied at a school for vocal training. The class in training was evaluated primarily by the period of time spent in studies of the individual groups under observation and then compared to professional singers. The residual pulmonary volumes were determined to be of highest importance in this assessment. This study then was correlated to experience with situations in which impairment of this pulmonary volume measurement existed, such as pulmonary asthma. The group of asthmatics in the singing profession was higher than that encountered in the average population group. The importance of these findings is primarily for the understanding of the physiology of the singing voice.

THURSDAY MORNING, 29 NOVEMBER 1979 BONNEVILLE ROOM 3, 9:00 A.M. TO 12:35 P.M.

## Session CC. Physical Acoustics V and Underwater Acoustics IV: Acoustic Remote Sensing III: Ocean Remote Sensing

Robert C. Spindel, Chairman

*Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543*

Chairman's Introduction — 9:00

### Contributed Papers

9:05

**CC1. Remote measurement of water currents using correlation sonar.** J. A. Edward (General Electric Co., HMED, Syracuse, NY 13221)

In correlation sonar the relative velocity between a source/receiver platform and an ensemble of scatterers distributed over the transmit beam pattern is obtained from the spatial and temporal cross correlations over an array of hydrophones. With a two-dimensional array and suitably encoded transmissions, it is possible to measure the three components of the mean-velocity vector plus the variation within the resolution volume of the radial component. For remote current profiling, one can measure the velocity components in a water volume bounded in angle by the transmit beam pattern and in range by the signal waveform and integration window parameters using a simple monostatic array configuration. If desired, a complete velocity profile can be measured during each transmission cycle. This paper will review the basic theory of correlation velocity measurement and will discuss system design constraints and operating parameters and their relationship to such performance measures as volume and depth resolution, profiling range, and measurement accuracy. In addition, results from a limited number of in-water measurements will be presented. [Work supported in part by the National Oceanic and Atmospheric Administration.]

9:20

**CC2. Doppler sonar measurements from FLIP.** R. Pinkel (University of California, San Diego, Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA 92152)

A pulse-to-pulse incoherent Doppler sonar has been developed at the Marine Physical Laboratory for use in the observation of oceanic

internal gravity waves. Mounted on the Research Platform FLIP, the sonar transmits at frequencies between 65 and 90 kHz at a peak power of ~32 kW. During preliminary tests, measurements have been made out to a range of 1.6 km, depths of 1.2 km, with a range resolution of 25 m. The sensitivity of the sonar measurements to a given internal wave group depends on both the orientation of the sonar beam and the direction of propagation of the wave packet. Since the vertical angle of wave propagation is a function of frequency, a predictable pattern can occur in sonar derived estimates of the internal wave frequency spectrum. These can be used to infer a limited amount about the azimuthal directional properties of the wavefield. A more direct approach is to use multiple sonars, operating simultaneously but pointed in different directions. A four sonar system is currently being constructed for use on FLIP. This should permit accurate estimates of azimuthal directionality of waves up to kilometer scales.

9:35

**CC3. Reflection coefficients of upper ocean sound-speed variability.** G. T. Kaye and J. Northrop (Code 5311, Naval Ocean Systems Center, San Diego, CA 92152)

Profiles of temperature and conductivity with depth were measured at two-minute intervals in June 1977 off R/P FLIP at a position 100 nautical miles west of San Diego. The profiler was developed by R. Pinkel of the Marine Physical Laboratory for upper ocean internal wave measurements. From these data, acoustic reflection coefficients were calculated for angles of incidence near normal for acoustic impedance changes on a one-meter vertical scale. Coefficients of these reflective layers, which ranged from about -75 dB to less than -110 dB, were observed to vary slowly in the presence

of high-frequency internal waves. It is suggested that acoustic remote sensing of these layers will be masked by biological reverberation interference. Comparisons between layer reflection coefficients and biological target strengths will be made to illustrate this point.

9:50

**CC4. Theoretical and experimental study of acoustic scattering in turbulent water.** T. J. Eisler and J. A. Clark (Mechanical Engineering Department, Catholic University of America, Washington, DC 20064)

The scattering of waterborne sound by turbulent flow from a jet is being examined. Turbulence is produced in this laboratory experiment by flow from a pressure vessel, through a 3-mm-diam tube, into a tank. Flow velocities out of the tube orifice in the tank are from about 1 to 10 m/s. The turbulent flow is insonified by tonebursts of about 500 ms duration from a 2.5-cm-diam, collimated acoustic beam. The region of insonification extends downstream from a point about 10-tube diameters away from the orifice. Acoustic frequencies are in the range from 500 kHz to 1 MHz. Scattered signals are detected by a directional hydrophone and a heterodyne receiver. Predictions of the acoustic scattering cross sections derived from theoretical models of turbulent flow will be presented and compared with experimental measurements. The dependence of scattering cross sections on flow velocity, acoustic frequency, bistatic angle, and the angle between the insonifying beam and the flow direction will be discussed. Doppler effects visible in the detected signals and anomalous scattering from vortex structures large compared to the acoustic wavelength will also be described. Implications of these results for underwater remote acoustic current sensors will be noted.

10:05

**CC5. Ocean acoustic monitoring and multipath stability.** R. C. Spindel (Woods Hole Oceanographic Institution, Woods Hole, MA 02543), J. L. Spiesberger (Scripps Institution of Oceanography, University of California, San Diego, La Jolla, CA 92093), and K. Metzger (University of Michigan, Ann Arbor, MI 48109)

A procedure for remote sensing of mesoscale ocean dynamic processes has been proposed by Munk and Wunsch [Deep Sea Res., **26A**, 123–161 (1979)]. It is based on the measurement of acoustic travel time variations between fixed sources and receivers distributed on the boundary of a large ocean area. Inverse theory is invoked to estimate the structure of the interior sound velocity field from travel time perturbations along individual ray paths. In this paper we present the results of a 48-day multipath stability experiment that demonstrates sufficient path stability to enable ocean acoustic remote sensing using this technique. A 220-Hz phase encoded signal with 64-ms resolution was transmitted at 10-min intervals between a moored source and a receiver at a 900-km range. Ray paths are clearly resolved, unambiguous, and identifiable. [Work supported by ONR.]

10:20

**CC6. Progress in remote sensing of temperature profiles in the atmosphere and oceans.** E. H. Brown (NOAA, Wave Propagation Laboratory, Boulder, CO 80302)

Encouraging results have been obtained with a method using three-dimensional interference patterns for the remote sensing of temperature profiles. Measurement of atmospheric wind (or ocean current) profiles are a by-product of the method. Latest experimental results and theoretical analysis of temperature and wind contributions are described. [Work supported by Army Research Office.]

10:35

**CC7. Acoustical observations of oceanic bottom boundary layer particulates.** John R. Proni (NOAA/AOML/Ocean Acoustics, 15 Rickenbacker Causeway, Miami, FL 33149)

Studies have been carried out on the oceanic bottom boundary layer using 20 kHz and 3 MHz sound systems. Data will be presented on the generation of a suspended layer of sediment, essentially confined within the bottom boundary layer, by nonlinear internal wave groups on the N.Y. Continental shelf in 120 m of water. Data will also be presented on the existence of a large underwater "cloud" of material found on the continental slope (water depth typically (600–300 m) which has been observed consistently over the last several years and which is highly acoustically reflective. Some data from a high resolution ( $\approx 1$  cm) 3 mHz-acoustical system on particulate concentration estimation will be presented.

10:50

**CC8. Acoustical studies of particulate dispersion in the ocean.** John R. Proni (NOAA/AOML/Ocean Acoustics, 15 Rickenbacker Causeway, Miami, FL 33149)

The question of accurate measurements of chemical or toxicant dilution rates in the ocean continues to be of fundamental interest to both the research and coastal management communities in the U.S. Results will be presented on the use of acoustical data to guide chemical sampling and reinforce chemical data on dilution rate estimates. Acoustical data will be presented which indicates the basic spatial variability scales of concentration of material dumped in the ocean for certain sites; the attendant implications for the statistical reliability of associated chemical data will be discussed.

11:05

**CC9. Backscatter from small-scale oceanic sound-speed structure.** G. T. Kaye (Code 5311, Naval Ocean Systems Center, San Diego, CA 92152)

Remote sensing of oceanic sound-speed structure with acoustic backscatter measurements near normal incidence differs greatly from similar atmospheric measurements because of masking effects in the water column due to biological reverberation. With difficulty this can be overcome if one can achieve detection via a large-aperture array. Here considerable processing gain is available if one steers for curved wavefronts in the array nearfield, taking advantage of the curvature differences between returns from specular reflectors and discrete point scatterers. The field data suggest that sound-speed structures provide returns which are well-correlated in phase, although signal amplitudes for reflectors are less than returns from the point scatterers. Additionally, the data show that these reflectors are highly directional with beamformed returns dropping more than 15 dB as the angle of incidence varied to  $2^\circ$  off from normal. These characteristics indicate that acoustic remote sensing of oceanic sound-speed structure with high-frequency narrow-beam echosounders will be difficult because of biological reverberation interference, the anisotropy of specular reflectors, and source motion. Examples of data taken with an 87.5-kHz echosounder off R/P FLIP will be shown for this purpose.

11:20

**CC10. Remote acoustic sensing of fluid and biological processes in the ocean.** M. H. Orr and F. R. Hess (Department of Ocean Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543)

A towed four-frequency acoustic backscattering system has been used to detect a variety of fluid processes in the ocean. A short review

of experimental results obtained during the past three years will be presented. Examples of acoustic records which show detected short period internal waves, mixed processes, the seasonal variability of the particulate distribution and dispersion of man-introduced particles in the ocean, and interleaving water masses will be presented. Internal wave spectra calculated from the acoustic records will be discussed. Acoustic data showing the interrelationship of predator and prey, and active biological avoidance of research instrumentation will also be presented. [Work supported by NORDA, NOAA and NSF.]

11:35

**CC11. Sodar observations in the Suez Canal zone.** G. Fiocco, G. Mastrantonio, and A. Ricotta (Istituto di Fisica, University of Rome, Rome, Italy and Laboratorio Plasma Spazio, Frascati, Italy)

The ship *SALERNUM* on its way to and from the Indian Ocean for a GARP assignment crossed the Suez Canal twice and was briefly stationed at terminal points of the Canal and in the Bitter Lakes. The ship carried a monostatic sodar with Doppler capability. During both passages which took place in January and March the sodar was operated almost continuously, and standard meteorological measurements were carried out. During the return passage a tethered balloon was deployed to provide temperature and humidity profiles, and a few pilot balloons were launched. The natural environment of the Suez Canal zone is somewhat unusual due to the presence of large and small bodies of water surrounded by desert. Very large spatial and temporal temperature and humidity contrasts exist which are not normally encountered elsewhere. A variety of boundary layer structures have been observed and analyzed.

11:50

**CC12. Marine acoustics in Australia.** N. Shaw, M. Marsden, and I. Bourne (Victorian Institute of Marine Sciences, 14 Parliament Place, Melbourne 3000, Australia)

A survey was conducted to identify groups, individuals, and institutions using acoustic probes in the marine environment and to ascertain (a) if the group's emphasis was research or industrial application of acoustics and (b) if their general field of interest was in objects and structure within the water column, bottom topography, or subbottom geological structure. Participants specified the acoustic equipment employed and assessed the system's performance, data quality, maintenance requirements, and general capabilities. Specific questions were asked about the group's requirements for remotely sensing physical parameters within the water column and detection of the related distribution of marine organisms. Relevant to this

joint JASA acoustic workshop was a call for greater interaction between researchers and those technologies now being developed separately for the atmospheric and aqueous media. For example, estimates of water turbulence and profiles of currents could be remotely sensed by acoustic techniques similar to those already developed for detecting atmospheric turbulence and wind profiles.

12:05

**CC13. An intercomparison of an acoustic remote current sensor and Aanderaa current meters in an estuary.** W. D. Scherer, K. A. Sage, and D. E. Pryor (Engineering Development Laboratory, C61, NOAA, Rockville, MD 20852)

An intercomparison experiment of a single-axis bistatic acoustic remote current sensor and Aanderaa current meters was conducted during the fall of 1978 in the Patuxent River near Solomons Island. The acoustic sensors were located on a platform at a mean water depth of 15 m and the acoustic axis pointed essentially downstream. The transmit frequency was set at 270 kHz, and the received signal was cabled to shore and heterodyned to 5 kHz. Up to 128 tone bursts of 10-ms duration were transmitted at 1-s repetition time every 15 min. The back-scattered volume reverberation data were analog recorded, subsequently digitized, and spectrally analyzed. The spectral estimates of the Doppler shift are derived for a number of range intervals and are compared with Aanderaa current speeds projected along the acoustic axis of the remote sensor. The comparison of the time series (15-min intervals) extends over a number of tidal cycles.

12:20

**CC14. A microprocessor-based MODEM for underwater remote sensing.** S. Prasad, B. B. Madan, and Navesh Kumar (Center for Applied Research in Electronics, Indian Institute of Technology, New Delhi, India 110029)

The underwater acoustic channel exhibits large dispersion of the signals in time and frequency domains. The dispersion is characterized by broadening and fading of the signals transmitted through it due to multipath propagation and random motion of particles constituting the medium. Conventional modulation/demodulation techniques, even with forward error correction and ARQ techniques and at moderate data rates, are quite ineffective for data communication over this channel. This paper briefly reviews some signalling schemes which might be suitable for the underwater acoustic channel. A comparative evaluation of them is also presented. The paper also describes a microprocessor-based instrumentation of a versatile MODEM which can implement these schemes under program control with a fixed amount of associated hardware.



**Session DD. Psychological Acoustics IV: Impaired Listeners and Effects of Noise Exposure (Poster Session)**

Roy W. Gengel, Chairman

*Department of Communication Disorders, University of Massachusetts, Amherst, Massachusetts 01003*

Chairman's Introduction—9:00

**Contributed Papers**

**DD1. Masking, temporal integration and speech intelligibility in subjects with noise-induced hearing loss.** Richard S. Tyler, Mariano Fernandes, and Elizabeth J. Wood (MRC Institute of Hearing Research, University of Nottingham, Nottingham NG7 2UH, England)

Three psychoacoustical measures of frequency analysis were obtained from ten normal-hearing subjects and twelve subjects with noise-induced hearing loss. First, thresholds of pulsed 500- and 4000-Hz pure tones were measured in the presence of a continuous, 60 dB/Hz spectrum level white-noise masker. Second, thresholds of pulsed, pure tones were measured in the presence of a continuous, 85-dB SPL tonal masker of 500 and 4000 Hz. Third, the intensity required to mask pulsed, 500- and 4000-Hz test signals at 10-dB sensation level was measured with sinusoidal masker of five adjacent frequencies [psychoacoustical tuning curves]. We also measured thresholds for 10- and 1000-ms duration signals of 500 and 4000 Hz as an indication of temporal integration. Two tests of speech intelligibility, one in a speech-spectrum shaped noise and one in a speech-babble background, were presented at a fixed intensity. Some hearing-impaired subjects clearly display increased thresholds in white noise (larger critical ratios), pronounced *downward* spread of masking, and flat or inverted tuning curves. Reduced frequency analysis and poor temporal integration were significantly correlated with reduced speech intelligibility in noise.

**DD2. Comparison of intensity discrimination in normal hearing, cochlear impairment and hearing loss simulated by masking.** M. Florentine (Room 36-761, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA 02139)

Intensity discrimination was measured as a function of level in normal hearing, cochlear impairment, and normal hearing with hearing loss simulated by masking. Both moderate flat losses and high-frequency sloping losses were studied. The stimuli were 500 ms tones with rise-fall times of 25 ms and interstimulus interval of 200 ms. Difference limens were based on several blocks of 100 trials using a two-alternative forced-choice procedure with feedback. In general, the high-frequency cochlear impairment and the high-frequency simulated loss show enlarged difference limens at levels below 90 dB SPL. However, one observer with a flat cochlear impairment showed enlarged difference limens throughout the entire intensity range, whereas the simulated flat losses show enlarged difference limens only in the range of recruitment. Interestingly, the difference limens are enlarged in the intensity range of abnormally rapid loudness growth. [Work supported by NIH.]

**DD3. Reaction-time measurement of tinnitus.** P. E. Goodwin (Department of Speech Pathology and Audiology, University of Denver, Denver, CO 80208) and R. M. Johnson (Kresge Hearing Research Laboratory, University of Oregon Health Sciences Center, Portland, OR 97201)

Reaction times (RTs) were obtained for hearing impaired subjects with tinnitus (experimental group) and normal hearing subjects with-

out tinnitus (control group) to the following sets of frequencies of the tinnitus patients: the tinnitus frequencies (which were associated with abnormal thresholds), nontinnitus frequencies associated with thresholds within normal limits, and nontinnitus frequencies with thresholds similar to the thresholds of the tinnitus frequencies. The only significant difference for RTs for the experimental group versus the control group was to the tinnitus frequencies. There were no significant differences between the two groups for RTs to the two sets of nontinnitus frequencies. There were no significant differences for RTs for the control group among the three sets of frequencies. However, for the experimental group, RTs to each of the three sets of frequencies differed significantly from each other. This study suggests that the measurement of RTs to auditory stimuli provides an objective measure of tinnitus, a means of comparing human and subhuman data, a method for determining recruitment, and a possible technique for mapping areas of tinnitus.

**DD4. Difference limen for amplitude compression of speech.** Igor V. Nábelek and Kazunari J. M. Koike (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37916)

It is postulated that the amount of information carried by speech can be altered by speech modifications which can be perceived aurally. Therefore, amplitude compression might increase intelligibility if a reduction of level differences between strong and weak speech sound is perceivable. Difference limen for attack ( $T_A$ ) and release ( $T_R$ ) times of compression for various compression ratios (CR's) based on a quality change was determined by the adjustment or ABX methods. A 1.7-s-long utterance with approximately constant maximal rms levels of words was submitted to wide-band compression. Masking noise added after compression at -10 dB (*re* speech) equalized the noise background for various conditions. The smaller the CR or the  $T_A$ , the shorter the  $T_R$  had to be to provide a perceptible change in quality; e.g., experienced normally hearing subjects needed  $T_R$  equal to or smaller than 420 and 250 ms to discriminate a quality change when CR = 5 and  $T_A$  equal to 10 and 3 ms, respectively, from linear amplification. Naive normally hearing listeners needed a much shorter  $T_R$  (between 30 and 40 ms for the same conditions). Testing of hearing impaired listeners is presently in progress. [Work supported by NIH NS 12946.]

**DD5. Low-level versus high-level amplitude compression.** Igor V. Nábelek (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37916)

Sensori-neural hearing-impaired individuals often exhibit a loudness function with a steeper slope at low levels than normally hearing people. At progressively higher levels the slope of the loudness function for the hearing-impaired approaches that of the normally hearing. Reducing the signal level differences at high levels (high-level compression, HLC) makes an abnormal loudness function, more so, while, reducing level differences at low levels (low-level compression, LLC) should result in a more normally

appearing loudness function. CVC words (MRT) were processed under three conditions: (a) only levels between  $L_r - 25$  dB and  $L_r - 13$  dB ( $L_r$  is the average maximal rms levels of the test words' vowels) were compressed (LLC) with a compression ratio CR = 4:1; (b) only levels above  $L_r - 13$  dB were compressed (HLC) with CR = 4:1; and (c) only levels above  $L_r - 25$  dB were compressed with the average compression ratio between  $L_r$  and  $L_r - 25$  dB in conditions "a" and "b" equal to 1.7. All three conditions had attack and release times of 1.5 and 30 ms, respectively. The performance of nine sensori-neural hearing-impaired subjects did not significantly differ among the three conditions investigated; this confirms Lippmann's (1978) findings with the compression system restoring normal equal loudness contours. [Work supported by NIH NS 12946.]

**DD6. Speech perception for various time constants of amplitude compression.** Igor V. Nábelek, W. Scott Wood, and Kazunari J. M. Koike (Department of Audiology and Speech Pathology, University of Tennessee, Knoxville, TN 37916)

Modified rhyme test words and CVC nonsense syllables were processed without or with background pink noise through a wide-band amplitude compressor set for a 2.5:1 compression ratio and various attack ( $T_A = 1$  through 42 ms) and release ( $T_R = 10$  through 370 ms) times. These stimuli were presented to nine subjects with sensori-neural hearing impairment. The compression threshold was set 25 dB below the average maximal rms level of the vowels in the test words or CVC syllables. The compressed signal was amplified with individually adjusted frequency responses and delivered to the listeners via a circumaural earphone. The presentation of speech was always at the same, comfortable level for each individual listener. The signal compressed without noise was sometimes mixed with a masking pink noise at  $-10$ ,  $-5$ , and  $0$  dB re speech. There were no significant differences in mean performances for different time constants, or between compressed and noncompressed stimuli. Performance slightly increased with compression, only when masking noise was added to the signal after processing ( $L_N - L_S \leq 5$  dB) and the attack and release times were shorter than 3 and 90 ms, respectively. [Work supported by NIH NS 12946.]

**DD7. A male-female speaker interaction for speech discrimination in noise with elderly compared to college age listeners.** R. W. Gengel and M. Marks (Department of Communication Disorders, University of Mass., Amherst, MA 01003)

Speech discrimination was measured in quiet and in noise for 30 elderly and 30 college-age listeners using a recorded male and female speaker. Results indicated there were no differences in intelligibility between speakers for the quiet condition. In contrast, for the noise condition, the male speaker was approximately 12% more intelligible than the female for the elderly group but approximately 16% less intelligible than the female for the younger group. This relative difference of about 28 percent in discriminability between talkers for elderly compared to college age listeners indicates a strong interaction between speakers and listener groups. Some statistical analyses of the audiometric data from the elderly listeners and electroacoustic analyses of the speech of the talkers will be discussed in an attempt to account for the interaction effect that was found.

**DD8. Prevalence of collapsible ear canals in an elderly clinical population.** Ronald L. Schow and Loren J. Randolph (Department of Speech Pathology and Audiology, Idaho State University, Pocatello, ID 83209)

Prevalence of collapsible ear canals was determined for 80 elderly (60-79 yr) subjects drawn for a clinical population. Pure tone air conduction thresholds were established at 500-4000 Hz using standard TDH 39 earphones mounted in (1) an MX-41/AR supra-aural

cushion and (2) an Eckstein, model CA circumaural cushion. A correction factor was established to insure equivalency between earphone cushions by testing 20 young adult subjects (20-29 yr). The criterion for a collapsible ear canal was a 15 dB or greater improvement in threshold at any one frequency when circumaural rather than MX cushions were used. None of the younger subjects showed this large a shift in threshold, but 29 elderly subjects (36%) did. These results indicate a higher prevalence for elderly persons than the 4% previously reported on a general clinical population [V. H. Hildyard and M. A. Valentine, Arch. Otolaryngol. 75, 422-423 (1962)]. Based on these data, measurement errors may cause inaccuracies in normative data of about 5 dB. It appears that circumaural cushions or bone conduction testing are needed to avoid errors in hearing tests on the elderly population.

**DD9. Temporal characteristics of the stapedial reflex with multiple sclerotic subjects.** M. A. Schechter, R. R. Rupp, H. B. Calder (Speech and Hearing Sciences Section, University of Michigan, Ann Arbor, MI 48104) and B. Milburn (Veterans Administration Hospital, Ann Arbor, MI 48105)

An impedance technique has been employed to quantitatively compare the temporal characteristics of the stapedial reflex in seven normal subjects and seven subjects with multiple sclerosis. Measures of latency and rise time of the reflex were collected in both groups. Analysis of the results revealed significant mean group differences for both the latency and rise time measures. Eliciting stimuli were tone bursts of 25 ms (2.5 ms rise/fall) and tones with a duration of 1 second (2.5 ms rise/fall) at both 10 and 20 dB SL. Three of the seven multiple sclerotic subjects displayed latency or rise time measures that were two or more standard deviations above the mean. The remainder of the pathological group fell within "normal" limits. Aberrant reflex characteristics would only be expected in those multiple sclerotic subjects with demyelination occurring within the auditory pathways. [Work supported by the Veterans Administration Hospital.]

**DD10. Further studies on the total-energy hypothesis.** W. D. Ward, T. Kiester, and C. W. Turner (Hearing Research Laboratory, University of Minnesota, Minneapolis, MN 55455)

Previous studies have established that in the chinchilla, the "critical level" for a 220-min continuous exposure to 700-2800-Hz noise is 105 dB SPL, in the sense that such an exposure produces neither significant permanent threshold shift (PTS) nor an increase in missing hair cells (MHC) over the normal median value of 50 to 100. However, 108 dB produced an MHC count of over 400, and 111 dB generated not only a 700-MHC count but also a 10-dB PTS at 2 kHz. Finally, 114 dB produced 1500 MHC and a 30-dB PTS at 2 kHz. Breaking the 114-dB exposure down to 22 10-min exposures administered twice weekly for 11 weeks reduced the MHC count to just over 300, with negligible PTS at 2 kHz, implying a reduction in "effective exposure level" of about 7 dB. Two more related exposures are reported here. A continuous 2200-min (1.5-day) exposure at 102 dB SPL produced 950 MHC and 10 dB of PTS, which are values that would be predicted on the basis of the total-energy concept—i.e., equivalent to 112 dB for 220 min. On the other hand, a 440-min exposure at 114 dB SPL using a pattern of 30 s on, 30 s off (cumulative exposure 220 min), also produced essentially the same effect as 112 dB for 220 min (800 MHC, 20 dB of PTS). The results thus affirm the validity of the total-energy principle in assessing single uninterrupted exposures to noise, as in acoustic trauma, but also indicate that rest periods, even short ones, reduce the permanent effect of repeated exposures. [Work supported by NIH grant NS 12125.]

**DD11. The effects of impulse noise level on the magnitude of asymptotic threshold shift (ATS) in the chinchilla.** R. P. Hamernik,

D. Henderson, and R. Salvi (Department of Otolaryngology, Upstate Medical Center, and Department of Mechanical Engineering, Syracuse University, Syracuse, NY 13210)

Four groups of monaural chinchillas were exposed to either 99, 106, 113, or 120 dB peak SPL reverberant impulse noise for 10 days. The impulses had a "B" duration of approximately 160 ms and were presented at the rate of 1 impulse/s. Hearing thresholds were monitored before, during, and for 40 days after exposure. After 24 h in noise, all four levels produced a stable ATS. The level of ATS at 0.5, 2.0, and 8.0 kHz grew at the rate of 2.5 dB for each dB increase in the impulse noise level. The PTS function at these frequencies suggests a threshold effect, i.e., the two lowest levels produced no PTS, while at the two highest levels PTS began to increase rapidly. The relation between ATS and PTS is complex. Both the 106 and 113 dB exposures produced similar levels of ATS but the 106 dB group showed no PTS while the 113 dB group developed as much as 30 dB PTS at 2.0 kHz. Although the data are limited, it appears that after some criterion level has been reached, PTS also grows at the rate of 2.5 dB/dB. The cochleograms for the four groups of animals show an orderly increase in the severity of damage. However, an orderly relation between individual cochleograms and audiograms was not found. [Work supported by NIOSH and NIEHS.]

**DD12. Predicting noise-induced threshold shift: A second look.** John Erdreich (Department of Otorhinolaryngology, University of Oklahoma, Oklahoma City, OK 73190)

Several years ago we presented a pilot study in which measurements of cubic and quadratic nonlinearities were compared with measurements of TTS<sub>2</sub> made on the same three listeners [J. P. Cobb and J. Erdreich, J. Acoust. Soc. Am. **60**, S104 (1976)]. From the data we suggested a relationship between threshold shift and amplitude nonlinearity. In an extended study we have examined this relationship at three frequencies separated by octave intervals. Temporary threshold shift at 1.4 times the exposure frequency produced by a 12-min 500-Hz, 1000-Hz, or 2000-Hz pure tone at 90 or 100 dB SPL was measured in ten listeners. The amplitude of the combination tone  $f_1 + f_2$  ( $f_2/f_1 = 1.2$ ;  $f_1 = 500, 1000, \text{ or } 2000 \text{ Hz}$ ) was measured for primary levels of 70 dB SPL. At 500 Hz the fatiguer produced little threshold shift. For the 1000-Hz exposures, the relation between TTS<sub>2</sub> and auditory nonlinearity is yet supported. At 2000 Hz, the threshold shifts produced were highly variable, possibly the result of system resonances. This will be explored further. Although these results are consistent with other studies suggesting a mechanical determinant of TTS susceptibility, additional evidence will be presented suggesting that there is also a metabolic component of susceptibility. [Supported by Dept. of the Army DAMD17-78-C-8074.]

**DD13. Threshold shifts in chinchillas exposed to octave-band noise centered at 125 Hz for three days.** Charles K. Burdick (Sensory Physiology, U.S. Army Aeromedical Research Laboratory, P.O. Box 577, Fort Rucker, AL 36362)

Maximum threshold shifts for exposures to octave bands of noise having center frequencies of 500–4000 Hz are found one-half to one octave above the exposure-band center frequency (Carder and Miller, J. Speech Hear. Res. **15**, 603–623). Maximum threshold shifts produced by exposure to octave-band noise centered at 63 Hz are found four to five octaves above the exposure-band center frequency. The question arises as to the frequency at which the maximum threshold shifts occur to octave bands of noise with center frequencies between 63 and 500 Hz. The present experiment was conducted to provide such information. A group of four monaural chinchillas was exposed for three days to an octave band of noise centered at 125 Hz at an intensity level of 110 dB SPL (94 dBA). Maximum threshold shifts occurred at four and four and one-half octaves above the center frequency of the exposure band, i.e., at 2000 and 2800 Hz.

**DD14. A critical level for auditory damage at high intensities.** G. Richard Price (U.S. Army Human Engineering Laboratory, Aberdeen Proving Ground, MD 21005)

Evidence from a number of different studies in which the ear was exposed to intense sounds can be interpreted as showing that there is a critical level ( $L_c$ ) above which the loss mechanism changes. Based on the research of Price [J. Acoust. Soc. Am. **66**, 456–465 (1979)], the physiological basis for  $L_c$  is thought to be mechanical stress at the level of the hair cell. When measured with conventional instruments,  $L_c$  depends on the spectrum of the sound as well as its specific pressure-time history. For an exponentially damped 1 kHz impulse,  $L_c$  has been calculated to be about 145 dB for the median human ear and about 155 dB for a rifle shot. For the cat ear,  $L_c$  may be about 20–25 dB lower than for the human ear. The presence of a spectrum-dependent critical level resulting from mechanical stress in the inner ear has significance for efforts to limit hazard from intense stimulation. It suggests not only how intense a sound is acceptable; but also how it should be measured. In addition, the presence of species dependent differences in  $L_c$  also has implications for use of animal models in noise effects research.

**DD15. Comparisons between the median hearing threshold levels for an unscreened black nonindustrial noise exposed population (NINEP) and several presbycusis data bases.** D. P. Driscoll and L. H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27650)

Hearing threshold level data representing an unscreened black NINEP have been collected. For comparison purposes the median threshold levels of previously established presbycusis data bases have been obtained from the literature. After each data set had been normalized relative to age 18, comparisons were made between the black NINEP and the presbycusis data base median hearing threshold levels for sex and age groupings. The unscreened black NINEP exhibits median hearing levels similar to those of the presbycusis data bases for ages less than approximately 45 years. However, for age groupings greater than 45 years the median hearing levels of the black NINEP are lower than those of the presbycusis data bases. [Work supported by the Rockefeller Foundation.]

**DD16. Effectiveness of three different types of ear protectors in preventing TTS.** L. H. Royster (Department of Mechanical and Aerospace Engineering, North Carolina State University, Raleigh, NC 27650)

The effectiveness of the E-A-R, Hear Guard, and Sigma insert hearing protectors in preventing daily temporary threshold shifts in two different industrial environments was investigated. The employees at both plant locations were exposed to an A-weighted  $L_{eq}$  of approximately 95.5 dB. The in-the-field protection provided by the E-A-R and Hear Guard ear protectors was investigated at one plant location and at a second plant location the effectiveness of the E-A-R and Sigma ear protectors was evaluated. For the two environments investigated the employees who wore the E-A-R ear protectors did not exhibit a significant TTS; however, the employees who wore the Hear Guard and Sigma style ear protectors did indicate a significant TTS. [Work supported by the Rockefeller Foundation (RF 78001), N.C. Department of Labor and the two industries that participated in the study.]

**DD17. An environmental stimulus for sudden death in infancy?** Philip Dickinson (Bickerdike Allen Partners, 442 North Main Street, Bountiful, UT 84010 and University of Utah, Salt Lake City, UT 84112)

The results of a detailed examination of the circumstances surrounding all cases of Sudden Infant Death (Crib Death) in Utah over a 3-yr period, strongly suggest the influence of an environ-

mental stimulus or trigger. Of the parameters investigated the leading candidate at the moment appears to be low-frequency noise—either naturally occurring in some sizes of family home, or produced in a moving automobile.

**DD18. Some differential effects of infrasound on man.** David Nussbaum (Institute for Aerospace Studies, University of Toronto, Downsview, Ontario M3H 5T6, Canada)

In this investigation 63 experimental and 21 control subjects were studied to learn more about realistic effects of infrasound on humans. Experimental subjects received infrasound at a frequency of 8 Hz and an intensity of 130 dB SPL, for a 30-min duration. Control subjects received identical sessions minus the infrasound. Physiological parameters measured included heart rate, respiratory rate, plethysmography, EEG, blood pressure, and audiograms. Performance variables consisted of time estimates and a short term memory task. Subjective response were evaluated via a mood scale as well as by verbal reports. Results indicate that individual differences play a role in the response to infrasound. Some experi-

mental subjects exhibited symptoms consistent with motion sickness and changes in their EEG patterns. Respiratory changes were also noted. A mechanism is suggested which might account for observed subjective response variations in light of physiological differences. These will be discussed in detail in the body of the paper.

**DD19. How loud are television commercials in comparison with the programs?** Richard C. Potter (Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138) and John F. Potter (Lincoln-Sudbury Regional High School, Sudbury, MA 01776)

The loudness of prime-time television programs and commercials was measured by obtaining 30-s  $L_{eq}$  sound levels at a typical viewer location in a representative living area. Preliminary results suggest that the overall sound levels are similar for both the commercials and the programs. However, the range of sound levels is much greater from the programs in comparison with the commercials, which all tend to be at the same higher sound level. The subjective differences are presented in terms of the statistical descriptions and frequency content of the sounds.

THURSDAY MORNING, 29 NOVEMBER 1979

GRAND BALLROOM III, 9:00 TO 11:53 A.M.

## Session EE. Speech Communication V: Acoustic Analysis of Speech

John F. Boehm, Chairman

*Department of Defense, Fort George G. Meade, Maryland 20755*

Chairman's Introduction—9:00

### Contributed Papers

EE1. Abstract withdrawn.

9:17

**EE2. Windowing in linear prediction analysis of voiced speech.** K. K. Paliwal and P. V. S. Rao (Speech and Digital Systems Group, Tata Institute of Fundamental Research, Homi Bhabha Road, Bombay 400 005, India)

The autocorrelation and covariance methods of linear prediction are used for pitch-synchronous and pitch-asynchronous analyses of voiced-speech segments and the effect of weighting the data, prior to its spectral analysis, by a tapered window function is studied. It is shown that for both pitch-synchronous and pitch-asynchronous analyses, the performance of the autocorrelation method improves considerably by the introduction of such a window function. In the case of the covariance method, no such windowing of data is required. Furthermore, it is shown that for pitch-synchronous analysis, the performance of the autocorrelation method is not much worse than that of the covariance method. Hence the use of the autocorrelation method is recommended over the covariance method if stability of the linear prediction filter is to be ensured.

9:29

**EE3. Estimation of short-time spectral envelope of voiced speech using maximum entropy method of spectral analysis.** K. K. Paliwal and P. V. S. Rao (Speech and Digital Systems Group, Tata Institute of Fundamental Research, Homi Bhabha Road, Bombay 400 005, India)

The Burg maximum entropy (ME) method of spectral analysis is used to estimate the short-time spectral envelope of the voiced speech signal. A number of vowel segments from continuous speech are analyzed using this method. The Burg ME method is compared with the autocorrelation and covariance methods of linear prediction using normalized linear prediction error and accuracy in estimating the speech spectrum as criteria. It is shown that the Burg ME method consistently gives better results than the autocorrelation method of linear prediction for both pitch-synchronous and pitch-asynchronous analyses. For pitch-synchronous analysis, this method performs slightly better than the covariance method, while for pitch-asynchronous analysis, its performance is marginally inferior to that of the covariance method.

9:41

**EE4. Improved estimate of vocal-tract areas from the speech wave.** Paul Milenkovic and Bishnu S. Atal (Bell Laboratories, Murray Hill, NJ 07974)

Areas obtained from linear prediction analysis of the speech wave often differ from actual voice-tract areas. Some of the differences are due to unrealistic assumptions made in the inversion procedure about the losses in the vocal tract, the spectrum of the glottal source, and the high-frequency spectrum of the vocal-tract impulse response. In this paper, we discuss the nature of errors introduced by these assumptions and describe a procedure for avoiding some of the errors. We discuss how lack of high-frequency spectral information creates ambiguities in mapping acoustic information to vocal-tract areas. Ambiguities are also caused by the uncertainty about the

source characteristics. We resolve these ambiguities by requiring that the recovered vocal-tract shape be smooth; the smoothness requirement ensures that the dimensionality of the area function is not increased in the process of specifying the missing information. Results obtained from tests on both natural and synthetic speech signals will be presented.

9:53

**EE5. Comparison of glottal pulse shapes derived from linear prediction residual and pole-zero model residual of speech.** B. Yegnanarayana (Department of Computer Science, Carnegie-Mellon University, Pittsburgh, PA 15213)

An important problem in voiced speech analysis is the determination of the characteristics of glottal pulses from speech signal. These characteristics can be derived in principle from the residual signal obtained by passing speech signal through the inverse of a model system for the vocal tract. The accuracy of the glottal pulse information thus derived depends on the accuracy of the model representing the vocal tract system. In this study glottal pulse shapes obtained by using an all-pole model and a pole-zero model are compared. The all-pole model is obtained from linear prediction analysis and the pole-zero model from a pole-zero decomposition technique. For most voiced sounds both the methods yield the same glottal pulse shapes. But for nasals and voiced fricatives the pole-zero model provides a more accurate characterization of glottal pulses than the all pole model. The results are explained on the basis of spectral approximation provided by the models.

10:05

**EE6. Evaluation of the effects of aperiodic components on a periodicity measure of voiced speech.** J. C. Bancroft (Department of Communication Research, National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, NY 14623)

A measure for quantifying the periodicity of normal and abnormal voiced speech was developed using the modified short time autocorrelation function. [J. C. Bancroft, J. Acoust. Soc. Am. Suppl. 1 **65**, S67 (1978)]. The aperiodic components within the voiced waveform are not evaluated with equal weighting when processed by this periodicity measure. The effects of shimmer, jitter, additive noise, and additive periodic noise were identified and their influence and contribution to the overall periodicity measure were evaluated. These results will be presented along with the effects of anti-aliasing filters on the measure.

10:17

**EE7. A metric for the height of certain pitch peaks in English.** M. Y. Liberman and J. B. Pierrehumbert (Bell Laboratories, Murray Hill, NJ 07974)

Subjects were asked to produce various utterances with a systematic variation of pitch range (thought of as "degree of overall emphasis"). These utterances each contained two main pitch accents; each utterance was considered as a possible answer to one of two different questions which effectively "foregrounded" either the first or the second pitch accent. We investigated the scaling of peak and valley  $F_0$  measurements as pitch range varied. Our conclusions: (1) for intonations of this type, there is a fixed, declining baseline independent of pitch range; (2) the equation,  $(FP - FB)/FB = k \times (BP - BB)/BB$ , fits our data quite well, where  $FP$  is "foreground peak,"  $FB$  is the "baseline value at the location of the foreground peak,"  $BP$  is "background peak," and  $BB$  is "baseline value at the location of the background peak." That is, the ratio of foreground peak to background peak is constant when they are expressed in terms of baseline units above the baseline.

10:29

**EE8. Vowel normalization and ear-canal resonance.** R. B. Monsen (Central Institute for the Deaf, St. Louis, MO 63110)

One male and one female subject spoke words of the form /h\_d/ containing each of the English vowels. The words were recorded simultaneously in two different conditions: (a) in the free field, and (b) through an ear of an acoustic manikin ("Kemar") fitted with a Zwislocki coupler. The recordings were transferred to disk for analysis by the Speech Microscope [Vemula *et al.*, J. Acoust. Soc. Am. **65**, S22 (1979)]. Each vowel was sampled at midpoint, and comparison was made of the waveforms and spectra in both recording conditions. The results indicate that, in Kemar's ear, the spectrum envelopes of front vowels have approximately the same shape above 2000 Hz for both male and female talkers, despite the fact that the actual formant frequencies may be very different. As Kemar hears it, the vowel /i/ has a single peak centered at 3000 Hz, which may correspond to  $F_2$  for females and  $F_3$  for males. For back and low vowels, the central peak in the spectrum is higher for females than for males. These results suggest that high-formant (front) vowels are partially normalized before they are "heard." [Supported by NINCDS Grant NS03856.]

10:41

**EE9. The intrinsic pitch of vowels in sentence context.** Christine H. Shadle (Electrical Engineering and Computer Department, Room 36-513, Massachusetts Institute of Technology, Cambridge, MA 02139), Janet B. Pierrehumbert (Bell Laboratories, Murray Hill, NJ 07974), and Mark Y. Liberman (Bell Laboratories, Murray Hill, NJ 07974)

It has long been accepted that high vowels have a higher intrinsic pitch than low vowels. First noted by J. W. Black [J. Speech Hear. Disorders **14**, 216-221 (1949)], this effect has been verified for different consonantal environments and in several languages. However, all studies of intrinsic pitch of vowels have used nonsense words, either in isolation or bearing the main phrasal stress in a carrier sentence. As a first step towards an understanding of how the intrinsic pitch of vowels interacts with intonation in running speech, this study examined the  $F_0$  of the vowels in "reed," "rod," and "rude" in sentence context. Results for four speakers are reported, for four locations in sentences matched for length and stress pattern. A statistically significant effect of vowel height was found for all speakers in all positions. In addition, /u/ was generally higher than /i/, confirming results reported by Lehiste and Peterson [J. Acoust. Soc. Am. **33**, 419-425 (1961)]. Possible explanations for these results will be discussed. [Work supported by NIH and Bell Laboratories.]

10:53

**EE10. Application of pitch perturbation measures to the assessment of hoarseness in Parkinson's disease.** C. Ludlow, D. Coulter, and C. Cardano (NINCDS, NIH, Federal Building, Room 1C-11, Bethesda, MD 20205)

To determine whether measures of pitch perturbation relate to the occurrence of hoarseness in a neurological disease, patients with Parkinson's disease (hypokinesia) and normal aging controls were recorded. Pitch period lengths were measured from recordings of extended vowel productions to compute: mean adjacent differences; mean deviations of cycles from predicted values based on adjacent cycles; and jitter ratios. Three trained judges rated hoarseness on a seven-point scale from the same recordings. Mean adjacent differences were linearly related to fundamental frequency in hypokinesia but not in normal speech. Judgements of hoarseness were significantly related to mean adjacent differences, mean deviations and jitter ratios in the hypokinetic speakers. No significant relationships between hoarseness and the perturbation measures were found in the aging controls. Judgements of hoarseness did not accurately

discriminate between normal and hypokinetic speakers, while the mean adjacent differences and deviation functions discriminated significantly between the normal and hypokinetic speakers. Mean jitter ratios did not discriminate normalcy from hypokinesia. Thus, adjacent differences and deviation functions are valid measures of hoarseness associated with hypokinetic dysarthria.

11:05

**EE11. Acoustic characteristics of first week infant cries and their relationship to sudden infant death syndrome.** R. H. Colton (Department of Otolaryngology and Communication Sciences SUNY-Upstate Medical Center, Syracuse, NY 13210) and A. Steinschneider (Department of Pediatrics, University of Maryland School of Medicine, Baltimore, MD 21201)

Several acoustic characteristics of cries produced in the first week of life by normal full term, premature, and SIDS sibling infants were analyzed using a sound spectrograph and a graphic sound level recorder. The characteristics analyzed were fundamental frequency, overall sound pressure level, duration,  $F_1$ ,  $F_2$ ,  $F_3$ , energy in the range from 0.05–4 kHz (spectral band #1), energy in range 4–8 kHz (spectral band #2), and energy in the band 8–16 kHz (spectral band #3). Of the three infant groups, normal newborn infants ( $N = 66$ ) exhibited the highest average  $f_0$ , shortest duration and lowest center frequency of the first three formants. In addition, most of the energy in their cries was below 8000 Hz. The premature infant group ( $N = 58$ ) exhibited greater variability among the measures and tended to produce greater energy in the higher frequencies. The group of SIDS siblings ( $N = 22$ ) exhibited the lowest  $f_0$ , highest  $f_0$ , and greatest energy in the higher frequencies of the cry. With respect to the cry variables, a multiple regression analysis revealed that the center frequency of the first formant was significantly related to the distinction between normal full terms and premature infants whereas fundamental frequency and the energy in the frequency band from 4–8 kHz were both significantly related to the normal full term/SIDS SIB distinction.

11:17

**EE12. Jitter and shimmer as physical correlates of roughness in sustained phonation—Re-examination.** Yoshiyuki Horii (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

Middle segments of sustained phonations [i, a, u] produced by 20 adult males were submitted to roughness magnitude estimation experiments by a panel of judges. The segments were also analyzed for mean fundamental frequency ( $f_0$ ),  $f_0$  standard deviation, jitter, and shimmer by an automatic  $f_0$  analysis program. The  $f_0$  analysis yielded an average jitter of 0.75% and an average shimmer of 0.38

dB. Multiple regression analysis showed that the roughness was most strongly related to the mean  $f_0$ . Contributions of jitter and shimmer to roughness, although statistically significant, were much smaller than expected from the near perfect correlations between roughness and jitter/shimmer reported by Wendahl (1966). Possible sources of this discrepancy and implications of the present findings are discussed. [This work is supported in part by the National Institute of Aging.]

11:29

**EE13. A comparison of the frequencies of the peak energy of the first formant ( $F_1$ ) of the spoken vowel  $\bar{a}$  for two occupations.** J. M. Jones (Human Factors Engineering, ADTC/SDEP, Eglin Air Force Base, FL 32542)

This study was based upon my hypothesis that a primitive sound language underlies human verbal language and communicates basic social, emotional, environmental, and task related parameters through physiologically encoded rules. A complicated thread of logic through the literature supported a prediction that the frequency of the peak energy of the first formant ( $F_1$ ) of the vowel  $\bar{a}$  would, therefore, be higher for a group of enlisted military personnel than a group of Ph.D. level managers or a group of nonmanaging engineers. The vowel  $\bar{a}$  was taped from 30 subjects, 10 from each group, and digitized. Using the fast Fourier transform, the amplitude frequency distribution was plotted. The  $F_1$  was significantly higher, as predicted, for the enlisted military group. This study was done as a masters thesis in psychology at the University of West Florida.

11:41

**EE14. Articulation loss of consonants using the English language.** R. H. Brey, W. Raventos, and K. Dick (Communicative Disorders Area, Brigham Young University, Provo, UT 84601)

Peutz [J. Audio Eng. Soc. 19, 915 (Dec. 1971)] reported a procedure for determining the transmission of auditory information in a room called articulation loss for consonants ( $AI_{\text{cons}}$ ). This procedure was an improvement over the articulation index, primarily because it included variables of room acoustics.  $AI_{\text{cons}} = (200 D^2 T^2 / V + a)\%$ , where  $T$  = reverberation time,  $V$  = volume of room,  $D$  = distance, and  $a$  = correction for the type of listener. Reverberation data were measured in constant percentage bandwidth using new microprocessor controlled instrumentation. Stimuli for his research were made up of Dutch consonant verbal consonants (CVC) words. Data will be presented to demonstrate the feasibility of using the same formula using English (CVC) words. Subjects were normal hearing individuals who were presented the materials under various reverberation times.

**Mini-Session on Lecture-Demonstrations in Acoustics**

Henry E. Bass, Chairman

*Department of Physics and Astronomy, University of Mississippi, University, Mississippi 38677***Chairman's Introduction—12:15*****Invited Paper***

Following presentation of the paper the psychoacoustics cassette tapes prepared at Harvard under the direction of David Green will be available for listening.

**12:20**

**Laboratory and demonstration experiments in acoustics.** Thomas D. Rossing (Department of Physics, Northern Illinois University, DeKalb, IL 60115)

The Acoustics Laboratory at Northern Illinois University has a library of 50 acoustics experiments, organized into 9 categories: 1. Motion, Energy, and Waves; 2. Perception of Sound; 3. Acoustics of Musical Instruments; 4. The Human Voice; 5. Electronic Production of Sound; 6. Acoustics of Rooms; 7. Electronic Music; 8. Environmental Noise; and 9. Other Experiments. These experiments are designed to correlate with several different acoustic courses, and therefore some of them can be performed at more than one level of sophistication or expanded into laboratory projects. Many of them can be performed, in whole or part, as demonstration experiments in the classroom. In this paper, a selection of experiments will be discussed, a few portable items of equipment demonstrated, and the author's thoughts on acoustics laboratory and demonstration experiments presented.

THURSDAY AFTERNOON, 29 NOVEMBER 1979      BONNEVILLE ROOM 1, 2:00 TO 5:05 P.M.

**Session FF. Musical Acoustics V: Organ Acoustics**

George R. Plitnik, Chairman

*Department of Physics, Frostburg State College, Frostburg, Maryland 21532***Chairman's Introduction—2:00*****Invited Papers*****2:05**

**FF1. Computer analysis of the effects of valve actions on initial transients in organ flue pipes.** Norman Kinnaugh (Department of Physics and Astronomy, Brigham Young University; Provo, UT 84602) and G. R. Plitnik (Department of Physics, Frostburg State College; Frostburg, MD 21532)

Computer measurements have been made of the onset transients of a set of experimental "Principal" organ pipes. The pipes were played individually from a number of valves closely resembling the two most common types of pipe organ valve actions. One action (representative of many electro-pneumatic types) has a small circular valve directly beneath the pipe; the other action (tracker type) consists of a long rectangular valve which admits air into a chamber which feeds the pipe. The analysis was a time-tracking correlation of the fundamental frequency from the time of valve opening through to the steady state. Partial tones were treated as a quasiharmonic series and analyzed similarly. Steady-state spectra of the same pipe played on different valve actions shows results which are contrary to that reported by some previous researchers. (The pipes and experimental wind chest were donated by the M. P. Moller Co., Hagerstown, MD.)

**2:35**

**FF2. The effects of mouth adjustments on the acoustic spectra of flue organ pipes.** A. W. Nolle (Department of Physics, The University of Texas at Austin, Austin, TX 78712)

The mouth height (cutup) of a flue organ pipe and the horizontal position of the upper lip relative to the axis of the jet have a great influence on both transient and steady-state waveforms. Effects of

these variables will be shown with data taken on adjustable pipes, open and closed, operating at 40 to 60 mm (water) wind supply pressure. The presentation concerns mainly (a) relative amplitudes of harmonic components in the steady signal, (b) noise components apparently created by the edgetone feedback process, and (c) multiphonic oscillation. The short pipe with highly anharmonic passive resonance frequencies is particularly discussed, because its harmonic content is due to the nonlinearity of the jet-to-pipe excitation described by Fletcher. Radiation from such a pipe may have second harmonic amplitude greater than 10% of fundamental amplitude, with the third harmonic much smaller. Significant characteristics of the jet-to-pipe excitation function are shown by the existence of a minimum of second harmonic content as the horizontal lip position is changed, and changes of the phase of the second harmonic relative to the fundamental.

3:05

**FF3. Some physical mechanisms underlying the pipe voicing art.** S. A. Elder (Physics Department, U.S. Naval Academy, Annapolis, MD 21402)

This paper examines underlying mechanisms for changes in the flue pipe spectra caused by selective changes in mouth cutup, pipe length, languid position, etc. The physics of the problem involves the mechanics of jet waves, and the subtleties of their interaction with sound waves. Special interest centers at lower lip, where jet waves originate, and at upper lip, where sound is produced by collision of jet and pipe fluid. The necessities of feedback amplification require jet and sound motions to act in a reciprocal way so that, for example, delay in impulse transmission along the jet is exactly matched by acoustic transmission delay in pipe. Actually, jet waves and their associated sound waves behave in remarkably self-consistent and complimentary ways. Just as sound waves travel up and down pipe to form discrete modes, jet waves also appear in discrete pairs, which travel up and downstream. One of the principal acoustic factors in voicing is the natural "stretching" of pipe mode frequencies, which tends to give pipes voiced to oscillate above resonance a different character from those voiced to oscillate below. An analogous property of jet is thickness Strouhal number, or nondimensional frequency. Since cutoff occurs at a critical Strouhal number, harmonic development on the jet is not possible unless  $S < 0.05 S_c$ . High and low Strouhal number jets, therefore, exhibit opposing spectral tendencies. Thus voicing is seen as adjustment toward optimum complementarity between jet and resonant cavity, consistent with a desired spectral output.

3:35

**FF4. Numerical method for calculating input impedances of organ reed pipes.** George R. Plitnik (Department of Physics, Frostburg State College, Frostburg, MD 21532)

The physical dimensions of organ reed pipe resonators are used to compute the input impedance as a function of frequency for each resonator. The numerically computed input impedances are compared to experimentally measured curves and some discussion of the tone color resulting from the interaction of a coupled reed-air column system is provided.

4:05

**FF5. Organ reed and air column regenerative relations: Tools of the voicer's trade.** A. H. Benade (Case Western Reserve University, Cleveland, OH 44106)

There is a sharp dynamical distinction to be drawn between an orchestral woodwind (in which the flow controlling reed is almost totally dominated by the resonances of its air column), and the human speech mechanism (where the frequency of the oscillating vocal folds is little affected by vocal tract resonances). The lip reed-air column interaction of the brass instruments has an intermediate position, but fairly close to that of the woodwinds. In the pipe organ one finds reed-air column combinations having all degrees of mutual influence, and problems sometimes arise when one member of the collaboration "escapes" from the instruction of the other. Over two decades ago work with Bruce Schantz to solve some pipe design problems, and to systematize voicing procedures (to control pitch, loudness, and tone color), introduced your speaker to a wide variety of easily reproduced, highly nonlinear, multidegree of freedom oscillations. Some of these will be discussed, and attention drawn to the way that they guided the development of our current understanding of orchestral wind instruments. [Work assisted by NSF.]

### *Contributed Papers*

4:35

**FF6. Temperament and tuning in organs with particular reference to accommodation variances applied to electronic divisions.** George W. Mulder (National Music Camp Division, Interlochen Center for the Arts, 4323 Lake Avenue, Interlochen, MI 49643)

Observations on current temperament and tuning practices of organs will be made. New combinations of pipes and electronic divisions in organs require a tuning accommodation in the electronic segment to accommodate temperature variance in the pipes. Through use of variable modulating devices the electronic unit may be readily



tuned to accommodate the variance in tuning of the pipes due to temperature differences. Additional reference to pitch variants within differing temperament systems will be discussed.

4:50

**FF7. Is unequal temperament coming back for pipe organs?** Donald E. Hall (Physics Department, California State University, 6000 J St., Sacramento, CA 95819)

For roughly a hundred years, equal temperament has been assumed practically without question for organs as well as pianos. But recently several organ builders have advocated unequal tempera-

ment, and put it into practice in their new instruments. Is this a minor aberration, or the wave of the future? First I will briefly review the unique acoustical problems presented by the pipe organ in choosing a temperament, and point out how past generations have coped with these problems. Second, I will report the results of a new survey of leading American organ builders. This will provide insight into the actual proportion of instruments now being unequally tempered, the reasoning and recommendations now being given by organ builders to prospective customers, and the prospects for unequal temperament to regain wide-spread importance. The author agrees with several of these builders that unequal temperament may deservedly have considerable importance and usage, yet without displacing equal temperament in the majority of church and concert-hall instruments.

THURSDAY AFTERNOON, 29 NOVEMBER 1979      BONNEVILLE ROOM 3, 2:00 TO 5:05 P.M.

### Session GG. Physical Acoustics VI: Miscellaneous

Alan B. Coppens, Chairman

*Naval Postgraduate School, Monterey, California 93940*

Chairman's Introduction—2:00

#### Contributed Papers

2:05

**GG1. Effects of finite aperture and nearfield oscillations in a parametric acoustic array.** Jacqueline Naze Tjøtta<sup>a)</sup> and Sigve Tjøtta<sup>a)</sup> (Applied Research Laboratories, The University of Texas at Austin, Austin TX 78712)

On the basis of a simplified analytical expression for the linearized sound field from a baffled piston source (see Paper GG2, by the same authors, in this Program), we calculate the effects of near-field oscillations on the generated difference frequency wave in a parametric acoustic array. We also derive a generalized aperture factor which is valid for short and long ranges. [Work supported in part by the Office of Naval Research.]

<sup>a)</sup> On leave from the Department of Mathematics, University of Bergen, Bergen, Norway.

2:20

**GG2. An analytical model for the nearfield of a baffled piston transducer.** Jacqueline Naze Tjøtta<sup>a)</sup> and Sigve Tjøtta<sup>a)</sup> (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712)

The linearized sound field from a baffled piston source (radius  $a$ , wavenumber  $k$ ) in a dissipative fluid is considered. A simplified parabolic equation is derived (for  $ka \gg 1$ ) and solved analytically. The solution matches a plane collimated beam in the vicinity of the source and has the Bessel function directivity in the farfield. The nearfield-farfield transition region is studied. The range of validity of the parabolic equation is discussed. Its exact solution is shown to be the first term of an expansion in powers of  $(ka)^{-2}$  for the solution of the Helmholtz equation. The higher-order terms are secular at distances of order  $a(ka)^{1/3}$  from the piston. The analytical results obtained for the linearized field can be used to calculate the effects of nearfield oscillations on nonlinear effects generated in soundbeams. (For example, see Paper GG1, by the same authors,

in this Program.) [Work supported in part by the Office of Naval Research.]

<sup>a)</sup> On leave from the Department of Mathematics, University of Bergen, Bergen, Norway.

2:35

**GG3. Axisymmetric propagation of a spherical  $N$  wave in a cylindrical tube.** Robert D. Essert and David T. Blackstock (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712)

Measurements have been made in air of the transient sound produced by a point source (an electric spark) inside a tube. Source and receiver are located on the tube axis, a distance  $L$  apart. In the received waveform the direct sound, an  $N$  wave, is followed by a sequence of reflected signals, the first representing a single reflection (at  $\frac{1}{2}L$ ), the second a double reflection (at  $\frac{1}{4}L$  and  $\frac{3}{4}L$ ), and so on. The first reflected signal resembles an  $N$  wave, but the second is a U-shaped pulse, the third an upside down  $N$ , the fourth an upside down U, and the fifth an  $N$  wave again. The four-waveform cycle is repeated indefinitely. The sequential change in waveform is attributed to the two-dimensional focusing (which causes a  $90^\circ$  phase shift) that occurs each time reflected rays cross the tube axis. Non-linear behavior has also been measured: when the source amplitude is increased, the phase shift of the reflected signals changes. Theoretical work, in which absorption is accounted for, is in progress. [Research supported by ONR.]

2:50

**GG4. Transmission loss measurement on a sheet of falling water.** D. R. Russell, D. F. Krafft, and C. J. Hurst (Department of Mechanical Engineering, Virginia Polytechnic Institute and State University, Blacksburg, VA 24061)

Transmission loss measurements were made on a sound barrier consisting of a thin sheet of freely falling water. These measurements were undertaken to see if such a sheet of water could be used for noise control where appropriate. The sheet of water was formed across a 2-ft-square duct connecting reverberant rooms. The intensity incident on this sheet of water and the intensity of the sound transmitted through it were measured using a two microphone technique. The transmission loss results thus obtained were checked using conventional reverberant room techniques. The measured transmission losses were lower than predicted, based on assumptions that the sheet of water acted as a point reacting mass wall with negligible fluid velocity. [Work supported by South-eastern Poultry and Egg Association.]

3:05

**GG5. Phase insensitive frequency tracked tone-burst spectroscopy.** John H. Cantrell, Jr., Joseph S. Heyman, and William P. Winfree (NASA-Langley Research Center, Hampton, VA 23665)

Ultrasonic spectra obtained by conventional Fourier-transformed pulse spectrum analysis techniques are influenced (1) by the shape of the electrical drive pulses used to generate the ultrasonic waveform and (2) by the sensitivity of the receiving transducer to phase front variations of the incident wave across its face. A new method of ultrasonic frequency analysis is introduced which combines the newly developed phase insensitive acoustoelectric transducer (AET) with a frequency tracked tone-burst spectroscopy (TBS) technique. TBS eliminates pulse shape artifacts in the ultrasonic spectrum by generating spectra equivalent to exciting the transducer with a true delta function shock excitation. Spectra from the AET-TBS combination are shown to produce clean spectral information even when sample inhomogeneity and geometrical irregularity severely degrades the spectral output obtained with conventional methods.

3:20

**GG6. A formal operator solution for the scattering of sound by sound.** Harvey C. Woodsum (Raytheon Company, Submarine Signal Division, P.O. Box 360, Portsmouth, RI 02871) and Peter J. Westervelt (Department of Physics, Brown University, Providence, RI 02912)

A general operator solution to the nonlinear acoustic wave equation for a viscous medium,

$$\left[ \nabla^2 \left( 1 + \rho_0^{-1} c_0^{-2} \eta \frac{\partial}{\partial t} \right) - c_0^{-2} \frac{\partial^2}{\partial t^2} \right] P = Q = -\beta \rho_0^{-1} c_0^{-9} \frac{\partial^2}{\partial t^2} (P^2)$$

is obtained from perturbation series,

$$P = \sum_{N=1}^{\infty} \epsilon^N P^{(N)}; \quad Q = \sum_{N=2}^{\infty} \epsilon^N Q^{(N)}.$$

The exact solution for  $P^{(N)} \sim O(\epsilon^N)$  is shown to be

$$P^{(N)} = \left( 1 - \exp \left\{ \left[ \left| \frac{\partial}{\partial t} \right| |c_0 \nabla|^{-1} \left( 1 - \frac{1}{2} \eta \rho_0^{-1} c_0^{-2} \frac{\partial}{\partial t} \right) - 1 \right] \mathbf{r} \cdot \nabla \right\} \right. \\ \left. \times \exp[-\text{Re}(\mathbf{r} \cdot \nabla)] \right) \left[ \nabla^2 \left( 1 + \rho_0^{-1} c_0^{-2} \eta \frac{\partial}{\partial t} \right) - c_0^{-2} \frac{\partial^2}{\partial t^2} \right]^{-1} \cdot Q^{(N)}.$$

This solution is a formal extension of Westervelt's 1972 non-singular operator solution [J. Acoust. Soc. Am. 59, 760 (1976)] to the case of a medium with viscosity. The evaluation of the operator expression above is shown to be straightforward, provided the primary field  $P^{(1)}$  is first expanded into a Fourier superposition of plane waves. Parametric array applications are discussed.

3:35

**GG7. Some comments on the interaction of sound with sound.** A. L. Van Buren, Peter H. Rogers (Naval Research Laboratory, Underwater Sound Reference Detachment, P.O. Box 8337, Orlando, FL 32856), and Jean Piquette (Stevens Institute of Technology, Hoboken, NJ 07030)

The fundamental equations and several derived equations governing the interaction of sound with sound are examined and compared in light of apparent discrepancies among published results for the scattering of sound by sound and the parametric receiver. The discrepancies are only significant for large angles between the interacting waves (such as are involved in the cross-beam experiment and the front-to-back ratio for the parametric receiver) and originate from inaccurate treatment of the problem. It is shown here that an accurate treatment requires that D'Alembertian terms be retained, that boundary condition be taken into account properly, and that the distinction between Lagrangian and Eulerian coordinate frames be maintained.

3:50

**GG8. On the angular spectrum of nonlinearly interacting acoustic waves.** F. H. Fenlon (Department of Engineering Science and Mechanics, and the Applied Research Laboratory, The Pennsylvania State University, State College, PA 16802)

In this paper the radiation field of a propagating finite-amplitude disturbance in an isotropic, homogeneous fluid is derived via spatial Fourier transform theory from the angular spectrum of plane waves in the wavenumber-frequency domain. By invoking the Fresnel approximation this field is shown to be equivalent to that obtained from the ZK equation [E. A. Zabolotskaya and R. V. Khokhlov, Sov. Phys.-Acoust. 15, 35-40 (1969)] thus leading to the important conclusion that if preferred one can bypass the latter and instead make use of the angular spectrum to evaluate the nonlinearly generated field. The paper concludes by using this approach to derive the plane wave angular spectrum of a parametric acoustic array and hence the difference-frequency field for the case of weakly interacting Gaussian beams.

4:05

**GG9. Abstract withdrawn.**

4:20

**GG10. Development of generalized nonlinear behavior of elastic waveforms in an arbitrary inhomogeneous, lossy medium.** G. R. Akers (Acoustic & Environmental Research Division, Naval Underwater Systems Center, New London, CT 06320)

A general method of developing the nonlinear wavefield for an arbitrary inhomogeneous medium incorporating energy loss mechanisms is presented. The equations of motion for an inhomogeneous medium with physical constants that are arbitrary

functions of position, are modified by the incorporation of a generalized loss function,  $L$ . This acts as a sink function by introducing a dissipative force field in the force balance equations that accounts for losses in the medium. A general power series solution is developed and the nonlinear wavefield solution is presented that incorporates the effects of losses and higher order interactions in its functional form. The effect of loss on the dispersion relationship is seen as a frequency shift of the wave and loss effects introduce harmonics which lead to waveform distortion. The classical theory of attenuation is extended by this treatment. [Work supported by Navy Long-term Training Committee.]

4:35

**GG11. A new approach to the solution of eigenvalue problems in circular flow ducts.** Sung-Hwan Ko (New London Laboratory, Naval Underwater Systems Center, New London, CT 06320)

Sound attenuation in acoustically lined circular ducts with fluid flow was investigated using a Taylor series method for the solution of the eigenvalue problem. The governing differential equation was numerically integrated across the duct cross section from the origin to the duct wall. The singularity of the differential equation at the origin was treated by analytically solving the differential equation valid only at the origin. Using values of the function and its derivatives at the origin, successive approximations of the eigenvalue were performed until the boundary condition at the duct wall was satisfied. Eigenvalues obtained by this method were compared with those obtained by a different method, and they were found to be in excellent agreement.

4:50

**GG12. Acoustically induced turbulence and shock waves in resonance tube.** Kiang H. Chou, Paul S. Lee, and David T. Shaw (Laboratory for Power and Environmental Studies, State University of New York at Buffalo, Buffalo, NY 14214)

Experimental investigations of acoustically induced turbulence and shock waves in a resonance tube are performed. Frequency ( $f$ ) and intensity ( $I$ ) effects of the acoustic field on turbulence and shock waves are studied using hot-film anemometer and FFT data processing unit. Measurements were made at the tube center and at approximately 1 mm from the wall at locations corresponding to loops and nodes of the standing waves. Sampled data were first conditioned then processed to estimate the characteristics of turbulence (Taylor microscale, the integral scale, autocorrelation, cross-correlation, and power spectrum density). It is found that for sound pressure level (SPL)  $\geq 162$  dB and frequencies within a narrow band ( $\sim 20$  Hz) around the resonance frequency, shock wave formation appears. Turbulence measurements were performed over a frequency range of 680–2740 Hz, with intensity over a range of 1.1–3.8 W/cm<sup>2</sup>. Below  $I = 1$  W/cm<sup>2</sup> no turbulent bursts are found. The turbulent power spectral density  $F$  and the wavenumber  $k$  are found to satisfy a power law  $F \propto k^\alpha$  with  $\alpha \approx -1.6$  to  $-2.1$ . The rms turbulent velocity  $u$  is experimentally found to have a  $I^{1/2}$  dependence, yet relatively insensitive to variation of  $f$ . Throughout the whole measuring range of  $f$  and  $I$ , the rate of energy dissipation per unit mass  $\epsilon$  is estimated to be  $10^6 \sim 10^7$  cm<sup>2</sup>/s<sup>3</sup>. Special emphasis on  $\epsilon$  is made as it vastly affects the agglomeration rate of the aerosol in the physiochemical processes.

THURSDAY AFTERNOON, 29 NOVEMBER 1979

GRAND BALLROOM I, 2:00 TO 4:50 P.M.

## Session HH. Psychological Acoustics V and Speech Communication VI: Recent Developments Re Hearing Impairment: A Tutorial Review

Lois L. Elliott, Chairman

*Program in Audiology and Hearing Impairment, Northwestern University, Evanston, Illinois 60201*

Chairman's Introduction—2:00

### *Invited Papers*

2:05

**HH1. Medical management of hearing disorders.** Jack D. Clemis, M.D. (Departments of Otolaryngology and Maxillofacial Surgery and Communicative Disorders, Northwestern University, Chicago, IL 60611)

The auditory system consists of an outer, middle, and inner ear, the auditory nerve, and central auditory pathways. Each area has specific functions and each has diseases that impair that function. The purpose of this discussion is to present some of the more common diseases that afflict the auditory system and to discuss current techniques in diagnosis, medical/surgical treatment and unsolved problems that require continued research efforts.

2:45

**HH2. Language learning in deaf children: The critical issues.** Rachel I. Mayberry (Program in Audiology and Hearing Impairment, Northwestern University, Evanston, IL 60201)

Research in deaf children's language has centered on three major areas: (1) the effects of educational method, parental communication, and hearing status on deaf children's acquisition of spoken language; (2) the way in which deaf children's spoken language deviates from that of normally hearing children; and (3) the way in which deaf children acquire sign language or develop gesture in its absence. This paper briefly reviews recent research in each of these areas in terms of critical questions germane to deaf children's language and also to first and second language learning.

**HH3. Psychoacoustic measures of hearing impairment.** Frederic L. Wightman and Israel Raz (Program in Audiology and Hearing Impairment, Northwestern University, Evanston, IL 60201)

Several laboratories have used psychophysical methods to study various aspects of hearing impairment in the belief that rigorous stimulus and response control would lead to stable and revealing results. Unfortunately, the data obtained thus far in our laboratory are disappointing. In spite of efforts to obtain homogeneous groups of subjects (using audiometric profiles), to practice them thoroughly, and to conduct rigorous, well-controlled research, the results of most of our experiments have been just as variable as the previous work. We feel this is probably due to the failure of standard audiometric measures to classify hearing-impaired listeners in terms of their basic auditory abilities. Current work in our laboratory is directed at developing alternative means for classifying hearing-impaired listeners, based on cluster analysis of performance on a large number of psychoacoustic tests. [Work supported by NIH Grant No. NS12045 and NS07108.]

**HH4. Recent developments in hearing aid selection methods.** Gerald A. Studebaker (Memphis State University, Memphis, TN 38105)

During the past 30 years, research on hearing aid selection methods has been largely confined to evaluation of the speech-discrimination-test-based method first proposed by Crichton in 1946. Recently, however, an increasing number of researchers, and clinicians, alike, have concluded that a new iconoclastic and broad-ranging look at this problem area is required. Many of the proposals made recently are, in fact, based on proposals first made in the 1930's and 1940's. The new proposals and their origins will be reviewed along with developments of the past ten years that enhance the likelihood that one or more of these new methods will soon replace the speech-discrimination-test-based approach as the "standard" method for selection of hearing aids for the individual user.

THURSDAY AFTERNOON, 29 NOVEMBER 1979      BONNEVILLE ROOM 4, 2:00 TO 5:15 P.M.

## **Session II. Shock and Vibration I: Instrumentation for Measurement and Analysis; Vibration Isolation**

Donal Maxwell, Chairman

*David W. Taylor Naval Ship Research and Development Center, Annapolis, Maryland 21402*

**Chairman's Introduction—2:00**

### ***Invited Papers***

**III. Measuring frequency response functions on operating systems.** Anton C. Keller (Spectral Dynamics Corporation, P.O. Box 671, San Diego, CA 92112)

In studying dynamic behavior of mechanical systems very often a three-step approach is used. In this classic approach, Step 1 involves analyzing actual running characteristics of a machine, structure, or process. Step 2 involves the determination of frequency response functions between various locations on the test system. Step 3 may then involve extraction of specific characteristics of the system such as structural mode shapes, damping parameters, stiffness values, or system resonant frequencies. Often it is quite possible to extract a good understanding of possible system design problems strictly from an analysis of operating data. However in most cases, the second step will usually involve a measure of selected frequency response functions through the use of external stimulus. In the past, this has almost certainly meant that the operating system would have to be shut down and basically be in a "static" state before artificial excitation could be furnished and the frequency response function measured. This has sometimes been impractical for several reasons. First, there are cases when a system must actually be operating before its real dynamic characteristics can be extracted. This would be the case, for example, if fluid film bearings represent a part of a total system dynamic. On the other hand, system operating characteristics may interfere with the very frequency response function which is to be determined. This could be the case in a rotating turbine blade where forced vibration responses would always be present in the output or in the case of a biological system measurement which might continually include the effects of heartbeat. In each of these cases, the system must continue to operate during a frequency response measurement but it is somehow necessary to eliminate the operating characteristics from measured frequency response functions. This paper

examines the classical techniques which have been used to extract frequency response information and some of the reasons why it has not been possible or convenient to eliminate running data from the desired measurement. An alternative approach is then given which has now been implemented in hardware and which offers a significant signal-to-noise enhancement over traditional techniques making it entirely feasible to extract complete frequency response information from operating dynamic systems.

2:27

**II2. Solving vibration and failure problems in prototypes.** Edward L. Peterson and G. W. Knobeloch (Computer Services, Structural Dynamics Research Corporation, Milford, OH 45150)

In the early 70's, modal analysis was introduced as a powerful new tool for solving vibration problems. Animated mode shapes provided new insight into the dynamic behavior of mechanical structures. These tools gave the engineer guidance in how to modify designs to improve dynamic performance. But, to see the effects of proposed design modifications, prototypes had to be altered and reanalyzed. Now, technology has taken us one step further; recent software developments help us to foresee how proposed changes in the prototype will affect dynamic performance. The data generated by modal analysis is used to construct a system model. Then we analyze altered versions of this model. Thus, with this new software, the designer can choose the best of several alternatives before he physically alters the prototype.

2:49

**II3. The intelligent FFT system.** Richard S. Rothschild (Nicolet Scientific Corp., 245 Livingston Street, Northvale, NJ 07647)

Vibration spectrum analysis is now performed using FFT computing devices almost exclusively. The digital nature of the FFT instrument has led to the incorporation of additional computational capability to enhance processing and interpretation. Much greater capabilities are becoming available through the marriage of computers and FFT analyzers, the most advanced of which are for modal analysis. The power and potential of a general or special purpose computer in combination with an FFT instrument-type analyzer is discussed with respect to typical vibration and acoustic problems that can be solved.

3:11

**II4. The impact of new technology on instrumentation for shock and vibration measurement.** William C. Huber (GenRad, AVA-West Division, 2855 Bowers Avenue, Santa Clara, CA 95051)

The dramatic evolution of new acoustic and vibration instrumentation is being driven by integrated circuit technology. Development of new microprocessors and analog processing circuits has enhanced the current generation of instrumentation, and in the next few years this trend will continue. Additional capabilities and new measurement techniques are being added rapidly. Several system architectures and developments in the area of FFT (Fast Fourier Transform) and digital and analog filtering are described.

3:33

**II5. Measurement and analysis of the dynamics of mechanical structure.** James O. Litz (Hewlett Packard, Santa Clara Division, 5301 Stevens Creek Boulevard, Santa Clara, CA 95050)

Modern methods for testing, modeling, and analyzing the dynamics of mechanical structures are given. Modes of vibration are shown to be the link between a model developed using finite-element methods and a transfer-function model, the elements of which can be measured in the laboratory. Testing techniques are reviewed with emphasis on the transfer function method.

3:55

**II6. Maximizing shock response spectra by wavelet parameter adjustment.** Theron Usher, Jr. (Unholtz-Dickie Corp., 3000 Whitney Avenue, Hamden, CT 06518)

The Time/Data TDV-20 is one of a number of shock synthesis systems used with power amplifier-shaker systems to provide specified shock response spectra (SRS). The TDV-20 drives the power amplifier with a signal containing a number of finite-length, weighted-amplitude, sinusoidal wavelets. The center frequencies, lengths, and relative time delays of the wavelets can be specified by the operator. Selection of these parameters within the limitations of other system constraints is discussed so that the SRS measured at the shaker table may be maximized.

**II7. Active vibration isolation systems.** J. N. Wilson (Division of Control Engineering, University of Saskatchewan, Saskatoon, Saskatchewan, Canada, S7N 0W0)

The need for improved isolation for operators of off-road vehicles is discussed. Referring specifically to agricultural vehicles, a review is made of some of the approaches taken in North America and Europe to improve seat suspension systems. Data comparing the performance of active and passive systems is presented. The presentation will be supplemented with slides and film strips.

### *Contributed Papers*

4:45

**II8. Electromagnetic noise in large induction motors.** Takayuki Koizumi and Masao Narita (Mechanical Engineering Department, Central Research Laboratory, Mitsubishi Electric Cooperation, Amagasaki, Japan)

An approach to prevent noises of large induction motors which are subjected to various forms of vibration caused by electromagnetic exciting forces is presented. From the results obtained in experimental investigations for many actual large motors, it has become clear that all the components of the noise and vibration paths from exciting point to radiational sound field must be taken into consideration for avoiding typical peak noise and vibration. A substructural system analysis is adopted to comprehend the vibrational behavior of all components by using FFT (fast Fourier transform) analysis and the transfer function method. The combined system simulation is done in a digital computer by using modal parameters to estimate the frequency response of the total structure. It means that the contributions of each component to the total noise and vibration are predicted without actual motors. Now, through this investigation, it is considered possible to predict quantitatively the effects of each component at least on vibration.

5:00

**II9. The forced vibrational response of a rectangular parallelepiped with stress-free boundaries.** Eric V. K. Hill (Department of Aerospace, Mechanical and Nuclear Engineering, University of Oklahoma, Norman, OK 73019)

Most of what is known about the nature of acoustic emission sources has been learned through the use of piezoelectric transducers coupled to rectangular parallelepiped or plate-type specimens. Unfortunately, the output of the piezoelectric transducer is not a true picture of the source signal; rather, it is the acoustic emission signal modified by the responses of the specimen, the specimen-transducer interface, and the transducer. As a first step in determining what acoustic emission looks like, this paper presents a closed-form normal mode solution for the forced vibrational response of a rectangular parallelepiped with stress-free boundaries. Knowing the specimen response will allow the calibration of piezoelectric transducers for all three wave types—dilatational, transverse, and surface waves—and this, in turn, will allow the characterization of acoustic emission sources.

THURSDAY AFTERNOON, 29 NOVEMBER 1979 BONNEVILLE ROOM 2, 2:00 TO 5:00 P.M.

### **Session JJ. Underwater Acoustics V: Propagation, Reflection, and Attenuation (Precis-Poster Session)**

John A. DeSanto, Chairman

*Naval Research Laboratory, Code 8160, Washington, D.C. 20375*

**Chairman's Introduction—2:00**

### *Contributed Papers*

2:05

**JJ1. A propagator matrix method for periodically stratified media.** Kenneth E. Gilbert (Naval Ocean Research and Development Activity, Sea Floor Division, NSTL Station, MS 39529)

The propagator matrix formalism is a convenient method for treating horizontally stratified media [F. Gilbert and G. E. Backus, *Geophysics* **31**, 326 (1966)]. Until now, however, it apparently has not been applied to periodic media. This paper shows how the propagator formalism can be used in conjunction with Sylvester's theorem to obtain an exact solution for the propagation of waves in a periodically stratified layer of arbitrary thickness. Explicit expressions are derived for the simple case of a periodically layered fluid. These expressions are used to investigate several interesting limiting

cases. Application of the method to more complicated cases is discussed.

2:09

**JJ2. Strength and diffraction parameters in long-range sound transmission.** Stanley M. Flatté (Division of Natural Sciences, University of California, Santa Cruz, CA 95064)

A method for calculating the strength and diffraction parameters,  $\Phi$  and  $\Lambda$  is described. Floquet's theorem is shown to be of use in the calculation of  $\Lambda$  for a range-independent sound-speed profile. The sensitivity of  $\Lambda$  and  $\Phi$  to changes in the sound-speed profile and the buoyancy-frequency profile is investigated. It is shown that

rays with large intensity tend to have small  $\Phi$  and small  $\Lambda$  except very near caustics. [Work supported by ONR Code 480.]

2:13

**JJ3. Sound-level fluctuations in a multipath environment.** J. A. Neubert (Code 5311, Naval Ocean Systems Center, San Diego, CA 92152)

For each array sensor  $j$  in a medium of  $M$  multipaths, the composite sound-level mean-squared sound-level fluctuation  $\langle \chi'^2 \rangle$  can be found by a complicated development to be constructed from the summation of the individual multipath mean-squared sound-level fluctuations  $\langle \chi'^2_{(m)} \rangle$  and of the multipath mean-square phase fluctuation  $\langle \theta^2_{(m)} \rangle$ , where  $\chi = \ln A$ ,  $A$  is the signal amplitude and  $m = 1, 2, \dots, M$ . Applying the appropriate fluctuation relations from Tatarski [*Wave Propagation in a Turbulent Medium* (McGraw-Hill, New York 1961)] and Schvachoko ["Sound fluctuations and random inhomogeneities in the ocean," *Sov. Phys-Acoust.* 13, 93 (1967)], the result is  $\langle \chi'^2 \rangle \approx 4M\bar{\mu}^2 L_n k_0^2 R$ , where  $\bar{\mu}^2$  is the mean-square sound-speed fluctuations,  $L_n$  is integral scale,  $K_0$  is the wavenumber and  $R$  is the range. [Work supported by NAVELEX Code 320.]

2:17

**JJ4. The effects of boundary condition approximations on coupled mode theory.** Steven R. Rutherford and Kenneth E. Hawker (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712)

This paper examines the effect of boundary condition approximations that are inherent to the coupled mode theories proposed by Pierce [*J. Acoust. Soc. Am.* 37, 19–27 (1965)] and Milder [*J. Acoust. Soc. Am.* 46, 1259–1263 (1969)]. The boundary condition approximation that is investigated involves the replacing of the continuity of the normal component of partial velocity boundary conditions with the continuity of the  $z$  component of partial velocity boundary conditions at sloping interfaces. This approximation is necessary in order to carry out the partial separation of depth and range variables effected in the mathematical formulation of the theory. This paper will show that a consequence of this approximation is that coupled mode theory applied to propagation media with nonhorizontal boundaries does not conserve energy. It will also be shown that a correction to coupled mode theory may be derived such that the proper boundary conditions are satisfied and energy is conserved, both to first order in the local bottom slope. Numerical examples will be presented which illustrate the nonconservation of energy effect and the proposed corrections to the theory. [Work supported by the Naval Ocean Research and Development Activity and the Naval Electronic System Command.]

2:21

**JJ5. The uniform asymptotic expansion with concave caustics.** Edward R. Floyd (Naval Ocean Systems Center, San Diego, CA 92152)

While the uniform asymptotic expansion had been developed for smooth convex caustics [D. Ludwig, *Commun. Pure Appl. Math.* 19, 215–250 (1966)], its validity to concave caustics has been tacitly assumed by the underwater acoustics community. A counter example based upon deriving the ansatz quantities from ray-tracing wavefronts for a concave caustic exhibits both a domain on the ensonified side of the caustic where the uniform asymptotic expansion is valid and the domain's complement where the uniform asymptotic expansion is degraded monotonically with penetration into this complement.

2:25

**JJ6. Comparison of range-dependent propagation models.** H. P. Buckner (Code 5311, Naval Ocean Systems Center, San Diego, CA 92152)

It is difficult to compare range-dependent propagation models because there are no known solutions that combine realistic bottom interaction and range changes. However, a necessary requirement of a useable model must be reasonable agreement between the model and exact solutions when there is no range dependence. To show this agreement, a four-step procedure is suggested: (1) range average of the sound levels to remove small scale fluctuations; (2) make contour plots of the difference in sound levels between the tests and exact solutions as a function of range and depth; (3) make continuous plots of the mean  $\pm$  standard deviation of the sound level difference averaged separately over near surface, mid-water, and near-bottom depths; (4) a written analysis by an experienced acoustician explaining the significance of the above data. These procedures will be demonstrated by comparison of FFT and Crank–Nicolson solutions of the parabolic equation with the exact normal mode solution for the case of liquid layer lying over a liquid half-space. [Work supported by NORDA, Code 500.]

2:29

**JJ7. Exact decomposition of bottom interacting paths via a generalized reflection coefficient expansion.** R. L. Deavenport, E. P. Jensen, and F. R. DiNapoli (Naval Underwater Systems Center, New London, CT 06320)

The exact range independent acoustic field for an arbitrarily structured multilayered medium may be expressed as a Fourier–Bessel integral involving generalized reflection coefficients. One obtains upon evaluation the total result due to all possible bottom interactions. It is desirable for some applications to decompose this total field result into path type contributions without incurring the approximations associated with ray theory. This is accomplished by first recognizing that the generalized reflection coefficient may be written in terms of partial local reflection coefficients. An infinite series of integrals results when the term in the kernel containing the local coefficients is expanded. Each term in this expansion then contains combinations of various local reflection coefficients which can be readily interpreted as the desired path contributions. The spatial amplitude and phase dependence is obtained by evaluating each term separately using the fast field program technique. Examples will be presented for a fluid subbottom structure having both constant and exponential depth dependent sound speed. [Work supported by NAVSEA 63D and NORDA 500.]

2:33

**JJ8. Born series sums for wave propagation in anisotropic, random media.** W. C. Meecham (UCLA, Los Angeles, CA 90024), M. T. Tavis, and H. T. Yura (Aerospace Corporation, P.O. Box 92957, Los Angeles, CA 90009)

The Born series for random wave propagation is considered for the physically interesting regime: the radiation wavelength short compared with the correlation lengths of the medium and the range large compared with such lengths. We treat a Gaussian medium with a Gaussian-form correlation function. These specializations lead to integrals for the moments with integrands which are themselves mainly of Gaussian form and can be performed algebraically using multivariate Gaussian distribution formulas. Specifically, for moments the, say, eighth-order Born term involves a 24-fold integration of which 16 can be done algebraically using the formulas. We are able for a two-scale anisotropic medium to sum the Born series for the first and second moments of the radiation field. The results reduce to familiar ones when we specialize to isotropic media. A novel characteristic is that for certain parameter ranges where the vertical medium scale  $L_v$  is very small compared with the horizontal scale  $L_H$  diffraction effects can substantially reduce the phase variance, thus reducing the randomizing effect of the medium on the propagated field. Anisotropy also leads to an intensity dependence on angle. The fourth-order moment, the

intensity correlation, is considered through fourth-order Born terms, it not being possible to sum the Born series in terms of simple functions for this moment. It is found that the propagating field is far from Gaussian for ranges of a few to many scattering lengths when diffraction effects are slight; specifically, when the Fresnel length for the vertical medium scale is much greater than the scattering length. On the other hand, the fourth-moment approaches the usually expected Gaussian form for moderate ranges in terms of scattering lengths when the Fresnel length is very small compared with that length. Validity ranges for the intensity correlation are established showing that summing through the fourth-order Born gives good results through part of the intermediate range preceding saturation of the field, depending upon the parameters discussed. For ranges where diffraction effects are not strong, the field is far from normal as judged by the intensity variance.

2:37

**JJ9. Transient wave propagation in bubbly liquids.** A. Bedford (ASE-EM Department, University of Texas, Austin, TX 78712), D. S. Drumheller, and M. E. Kipp (Sandia Laboratories, Albuquerque, NM 87115)

Solutions for the pressure profiles of transient plane waves in a bubbly liquid have been obtained by using a finite difference procedure. In contrast to previous analyses, the liquid was not treated as incompressible. However, relative motion between the bubbles and the liquid was neglected. Two waves were clearly observable in the solutions. The first was a precursor wave in the liquid which had very little effect on the pressure in the bubbles. It was followed by a second wave in which the compression or expansion of the bubbles took place. An important finding was that the compressibility of the liquid had a substantial effect on the pressure profile of the second wave, which implies that the liquid compressibility should be included in future analytical studies of transient pulses in bubbly liquids. Comparison of the numerical solutions with the results of shock tube experiments [L. Noordzij and L. van Wijngaarden, *J. Fluid Mech.* **66**, 115–143 (1974)] showed good agreement except that the solutions did not predict oscillations that were observed near the head of the pulse in the experiments. [Work supported by U.S. Department of Energy.]

2:41

**JJ10. The directional dependence of normal mode attenuation due to scattering from a rough ocean surface.** David M. F. Chapman (Defence Research Establishment Atlantic, P.O. Box 1012, Dartmouth, Nova Scotia, Canada, B2Y 3Z7)

Using the boundary condition for the coherent component of sound at a rough ocean surface due to Kuperman and Ingenito [W. A. Kuperman and F. Ingenito, *J. Acoust. Soc. Am.* **61**, 1178–1187 (1977)], the actual directional dependence of the normal mode attenuation coefficients can be determined for attenuation due to scattering from a rough ocean surface whose surface variance spectrum has a  $\cos^2\phi$  anisotropy. Each normal mode has a directional attenuation of the form  $a_n + b_n \cos 2\theta$ , with the azimuthal angle of propagation  $\theta$  being measured from the wind direction, and the coefficients  $a_n$  and  $b_n$  being calculated from the eigenfunction and eigenvalue of the  $n$ th normal mode and the radial portion of the variance spectrum, which is arbitrary.

2:45

**JJ11. Range-dependent propagation in coastal waters.** W. A. Kuperman and F. B. Jensen (SACLANT ASW Research Centre, La Spezia, Italy 19026)

The parabolic equation (PE) is used to study sound interaction with a sloping ocean bottom and some of the results are compared

to a model tank experiment. The bottom, as inputted in the PE model, is characterized by sound speed, density, and attenuation. Practical running times have been achieved for these coastal water studies by utilizing a real time computer system with an array processor. Up and down slope propagation is studied demonstrating mode conversion and mode cutoff. For the upslope propagation we find that each mode cuts off by radiating a discrete beam into the ocean bottom and that there is very little conversion into the next lower mode. Mode cutoff occurs at approximately the depths predicted by normal mode theory but the cutoff takes place over a finite distance which essentially provides an aperture of many wavelengths for radiation of the beam into the bottom. An important conclusion that one can draw from this study is that mode-coupling theories should also include the coupling into the local continuous modes which would then provide the description of the radiation into the bottom.

2:49

**JJ12. Numerical tests of parabolic equation algorithms for field computations over totally reflective three-dimensional bottoms.** Ronald L. Dicus and Evan B. Wright (Naval Research Laboratory, Washington, DC 20375)

Computations with two- and three-dimensional versions of the parabolic equation (PE) split-step Fourier algorithm were compared with normal mode theory and diffraction theory to ascertain the adequacy of PE computations for propagation over a totally reflective bottom. All PE computations were for 10 Hz in a constant sound speed water column with discrete depth and range steps of 20 and 100 m, respectively. Totally reflective bottom boundaries were simulated by a discontinuous fluid sound speed at grazing angles less than critical. Transmission loss over a flat  $13^\circ$  critical angle fluid bottom reproduced essentially the same structure as given by normal mode theory for ranges less than 200 km. PE computations of the field near the edge of a perfect reflector provided a reasonable approximation of the general features exhibited in diffraction theory results. Swept beam three-dimensional PE computations incorporating reflections from an arbitrarily inclined plane correctly displayed reflected beams at angles in agreement with geometrical considerations. Implications for computation of propagation effects over realistic bottoms composed of three-dimensional sediment or rock facets are discussed.

2:53

**JJ13. The effect of sediment rigidity on bottom reflection loss.** P. J. Vidmar (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712)

The effect of sediment rigidity on bottom reflection loss,  $RL = -20 \log_{10} (|R|)$  where  $R$  is the plane wave reflection coefficient, is studied using a computational model [P. J. Vidmar and T. L. Foreman, *J. Acoust. Soc. Am.* **65**, S43 (1979)]. A single inhomogeneous turbidite layer overlying a homogeneous substrate is treated with an emphasis on low frequencies (10–200 Hz) and low grazing angles ( $0^\circ$ – $45^\circ$ ). We find that sediment rigidity can be neglected for thick ( $\sim 500$  m) layers while it can produce significant increases in RL for thin layers ( $\sim 36$  m). The frequency dependence of RL for thin layers has a low frequency regime characterized by large peaks ( $\sim 25$  dB) recurring at short ( $\sim 2$  Hz) intervals. At high frequencies there remains a persistent increase in RL ( $\sim 4$  dB). A ray picture, based on an expansion of the reflection and transmission coefficient in the small sediment  $S$  wave speed, provides a basis for understanding these results. The most important mechanism for sediment  $S$  wave excitation is found to be compressional wave conversion at the sediment–substrate interface. [Work supported by the Naval Ocean Research and Development Activity and the Naval Electronic Systems Command.]



**JJ14. A different . . . view . . . of attenuation . . . Part III, Conclusion.** Louis C. Maples (Naval Underwater Systems Center, New London, CT 06320)

"A Different Point of View on the Role of Attenuation as a Component of Total Propagation Loss in Underwater Acoustics," a paper given at the 96th Meeting of the Acoustical Society, and its sequel, Part II, presented at the 97th Meeting, presented a mathematical model for propagation on the axis of a sound channel which proposed simple random-phase addition of multipath signals as the mechanism determining spreading loss, rather than the usually accepted cylindrical spreading. Interpretation of experimental attenuation data using the new expression results in lower coefficient values than those of Thorp, Browning *et al.*, based on the same data. Earlier studies involved eye-fitting to published plots, and least-squares fits to calculated propagation data produced by Weinberg's generic model. The present paper compares "old and new" attenuation model analyses of actual raw data, as planned for Part II, but made possible only in July 1979, with obtaining of the original records. [Work supported by the Naval Underwater Systems Center.]

3:01

**JJ15. Propagation of sound from water into a sloping fast bottom.** A. B. Coppens, J. V. Sanders, N. Bradshaw, M. Kawamura, and I. Ioannou (Department of Physics and Chemistry, Naval Postgraduate School, Monterey, CA 93940)

When sound is propagated from large distances toward the apex of a fluid wedge overlying a fluid bottom in which the speed of sound is greater, acoustical energy must be transmitted into the bottom as the various normal modes in the wedge are cut off. (Alternatively, we can say that the rays of sound, because of the continual increase in their grazing angles of incidence on the wedge-bottom interface resulting from successive reflections between the wedge surfaces, will eventually obtain angles of incidence greater than critical, at which time they will begin to be refracted into the bottom; this simple view is qualitative only.) In our approach we use the method of images to calculate the distribution of pressure and its phase at the interface, and then treat the interface as a distribution of sources and calculate the field radiated into the bottom by the Green's function integral. Both calculations require a high-speed digital computer. Preliminary results and comparisons with experiments will be discussed. [Work supported by ONR and NOSC.]

3:05

**JJ16. Three-dimensional ray trace studies over a rough ocean bottom topography.** G. R. Giellis (Naval Research Laboratory, Washington, DC 20375)

In sea areas where long-range acoustic propagations involves repeated bottom reflection, it is essential to consider the three-dimensional aspect of the reflection. Computational methods by DeWitt [J. Acoust. Soc. Am. **62**, S19(A), (1977)] have been extended to include horizontal changes in the sound speed profile and bottom loss at each reflection. It is now possible to study horizontal ray deflections in areas containing a mid-ocean ridge. An area of 120 000 km<sup>2</sup> of sea bottom is represented by triangular facets developed from a grid of bathymetric points at one km intervals. This completely describes the bottom to the degree that it is presently known from the best existing charts. Sound speed profiles can change at large zone boundaries but horizontal refraction is neglected. Bottom reflection losses are calculated as a function of grazing angle for each bottom bounce of the ray. Rays which have accumulated a loss greater than a selected threshold can then be discontinued. Illustration and conclusions of the study will be given.

**JJ17. Multipath analysis of explosive source signals in the ocean.** N. R. Bedford, S. K. Mitchell, and G. E. Ellis (Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78712)

A computer-based measurement system has been developed for the measurement of acoustic propagation loss in multipath arrivals of signals from explosive sources. The system involves both signal measurement operations and careful theoretical representations of the measured quantities. Initial applications have been for measurement of bottom reflection losses using signals which reflect many times from the bottom. It has been found that the levels of arrival pulses can be strongly affected by predictable surface reflection effects. Measurements demonstrate consistent bottom reflection loss values across homogeneous basin areas. [Work supported by Naval Ocean Systems Center.]

3:13

**JJ18. Optical simulation of acoustic propagation through deterministic and internal wave sound speed variations.** L. E. Estes and G. Fain (Department of Electrical Engineering, Southeastern Massachusetts University, North Dartmouth, MA 02747)

We make use of an argon-ion laser operating at 4880 Å to model portions of the 406-Hz MIMI experiments. Our simulation covers a range of 250 km. The Garret-Munk internal wave model is used to construct two-dimensional sound speed profiles at 15-min intervals. Each profile is recorded as a separate photographic plate. The plates are imaged using a high-intensity ruby laser beam to differentially heat a fluid. The argon-ion laser beam propagates through the light speed variations caused by the heating. The modeled rms sound speed fluctuation at the sound channel is approximately 0.01%. Phase and intensity measurements are made. Our results are compared with numerical models and extrapolated experimental results. Using the deterministic sound speed profiles reported by Clark and Kronengold, [J. Acoust. Soc. **56**, 171-1083 (1974)], we present measurements of intensity as a function of range and depth. [Work supported by the Office of Naval Research.]

3:17

**JJ19. Shadowing by seamounts.** H. Medwin and R. Spaulding (Physics and Chemistry Department, Naval Postgraduate School, Monterey, CA 93940)

Major topographical features such as seamounts can cause large propagation losses which are functions of sound frequency and the type of interruption of the sound path. A 2 × 2 × 0.2-m high model of Dickens Seamount has been constructed of wood and plaster to a scale of 1:7874 and the sound shadowing in air has been studied for various source/receiver locations and for five octaves of laboratory frequencies. A new universal concept, the farfield "diffraction strength," is defined. The diffraction strength obtained from laboratory measurements is used to predict the isovelocity, frequency-dependent diffraction loss at sea. This loss is added to predictions of ray refraction losses during propagation up to, and away from, the seamount. This sum agrees well with the total long-range-ocean-propagation loss measured beyond Dickens by G. R. Ebbeson, J. M. Thorleifson and R. G. Turner [J. Acoust. Soc. Am. **64**, S76 (1978)]. The scale model is a relatively inexpensive device to aid in the prediction of shadowing losses at sea.

3:21

**JJ20. Array measurements of long-range, low-frequency signal propagation in the seabed.** Arthur B. Baggeroer (Departments of Ocean and Electrical Engineering, Massachusetts Institute of Technology, Cambridge, MA 02139)

Multichannel arrays and high-resolution velocity analysis have been used to measure the characteristics of low-frequency (5–20 Hz) signals propagating over long ranges (10–100 km) in the seabed. The ability to resolve the various paths both spatially and temporally has provided unique measurements of (i) the partitioning of the energy in the paths versus range, (ii) the behavior near critical angles for both refracted compressional and shear waves, (iii) multipath effects due to local inhomogeneities such as seamounts, and (iv) multiple reflected signals that are more intense than their corresponding direct paths. Multichannel data from both thickly and thinly sedimented regions has been employed in making these measurements. [Work supported by ONR.]

3:25

JJ21. The structure of simple and composite convergence zones.  
H. M. Walkinshaw (Bell Laboratories, Whippany, NJ 07981)

Progressive stages in the development of RSR convergence zones were shown in the previous paper [“A Quantitative Definition of Depth Excess—How Much Is Enough?,” J. Acoust. Soc. Am. Suppl. 1 **65**, S15 (1979)] to form a convenient set of criteria for defining depth excess. This present paper uses both parabolic approximation and ray diagnostic codes to explore general properties of simple and composite convergence zones and to show how changes in the arrival structure of waterborne paths give rise to corresponding changes in sound transmission with range. The role of the surface grazing ray, retrograde-range smooth “caustic” ray, and recursive ray in determining RSR zone width and intensity are described. Similar ray analogs are developed for RR convergence zones with cusped caustics and for composite RR–RSR convergence zones. The effect of source/receiver geometry on zone structure asymmetry and zone width are demonstrated for shallow and deep sources with receivers at source depth and reciprocal (conjugate) depth. The supporting calculations were made at 100 Hz to 200 nmi for shallow (18 m) and deep (152 m) sources at four water depths (3840, 4145, 5030, and 5500 m).

THURSDAY AFTERNOON, 29 NOVEMBER 1979

BONNEVILLE ROOM 5, 2:00 P.M.

### Joint Meeting of Standards Committees S1 and S3

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

#### Meeting of Standards Committee S1 on Acoustics

D. R. Flynn, Chairman S1

*National Bureau of Standards, Environmental Noise Section, Washington, D.C. 20234*

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

#### Meeting of Standards Committee S3 on Bioacoustics

W. Yost, Chairman S3

*Parmly Hearing Institute, Loyola University  
6525 North Sheridan Drive, Chicago, Illinois 60626*

**Standards Committee S3, Bioacoustics.** The current status of standards under preparation will be discussed. In addition to those topics including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years.

**Session KK. Noise VI: Noise Generation and Propagation**

Richard M. Guernsey, Chairman

*Cedar Knolls Acoustical Laboratories, 9 Saddle Road, Cedar Knolls, New Jersey 07927*

Chairman's Introduction—9:00

**Contributed Papers**

9:05

**KK1. Dispersion of sound in a combustion duct by fuel droplets and soot particles.** J. H. Miles (NASA Lewis Research Center, 21000 Brookpark Road, Cleveland, OH 44135) and D. D. Raftopoulos (The University of Toledo, Toledo, OH 43606)

Dispersion and attenuation of acoustic plane wave disturbances propagating in a ducted combustion system caused by fuel droplet and soot emissions from a jet engine combustor are studied. The attenuation and dispersion are due to heat transfer and mass transfer and viscous drag forces between the emissions and the ambient gas. Theoretical calculations show sound propagation at speeds lower than the isentropic speed of sound at low frequencies. Experimental results are in good agreement with the theory.

9:20

**KK2. A time-dependent difference theory for sound propagation in ducts with shear flow.** K. J. Baumeister (NASA—Lewis Research Center, 21000 Brookpark Road, Cleveland, OH 44135)

A time-dependent numerical formulation is derived for sound propagation in a two-dimensional straight soft wall duct with a sheared mean flow. The time-dependent continuity and momentum equations are developed along with the soft wall boundary conditions. The appropriate governing equations and boundary conditions are solved simultaneously by an explicit iteration procedure. The analysis begins with a harmonic noise source radiating into a quiescent duct. This explicit method calculates stepwise in real time to obtain the transient as well as the "steady" state solution of the acoustic field. The von Neuman method is used to develop relationships which describe how sound frequency and grid spacing effect numerical stability. The soft wall boundary conditions require special treatment to keep the iteration technique stable. Example calculations are presented for sound propagation in hard and soft wall ducts. The time-dependent analysis has been found to be superior to the steady finite difference and finite element techniques because of much shorter solution times and the elimination of matrix storage requirements.

9:35

**KK3. Flyover noise of a wide body aircraft.** S. K. Lanter, B. N. Shivashankara, F. G. Strout and R. Ayyagari (The Boeing Company, Seattle, WA 98124)

Aircraft noise in the vicinity of airports is a well-recognized problem. Rules and regulations are in effect that set limits on the noise levels of aircraft. Although the airplanes in the commercial fleet meet the appropriate regulations, there is always a constant effort within The Boeing Company to lower the noise levels. This requires a thorough understanding of the source mechanisms and propagation effects. The total noise of an airplane is composed of engine noise and airframe noise. The engine noise itself includes various components such as jet, core, fan, and turbine noise. There

has been a lack of good quality flight noise data base, especially for wide body aircraft with high bypass ratio engines. To alleviate this, several test programs have been completed within The Boeing Company. The results of one of these programs, the test of a Boeing 747 airplane fitted with Pratt and Whitney JT9D engines, are presented in this paper. From the noise results, regions of jet noise dominance, fan noise dominance, and core noise dominance are identified. Methodology that will be used for more accurate breakdown of the noise components in regions where they are not dominant is outlined.

9:50

**KK4. Large scale model measurements of air frame noise using cross correlation techniques.** W. R. Miller, W. C. Meecham (UCLA, Los Angeles, CA 90024), and W. F. Ahtye (NASA—Ames Research Center, Moffett Field, CA 94035)

Cross-correlation techniques are used to measure the sound radiated by wing/flap airfoil configurations in the NASA—Ames 40- by 80-ft wind tunnel using a 5.8-m-high model, with three deployed flaps. The sound from flap corners exceeds other airframe noise by 10 dB and more; the noise from the leading, outbound corner of the leading flap seems to be the strongest. The sound is estimated using two formulas based on standard aeroacoustic theory and one method using the near-, farfield crosscorrelation; this last is essentially independent of such theory—all three are in fair to good agreement with one another. The classic dipole angular distribution pattern for one dipole is compared with measurements; it is found that there is qualitative but not quantitative agreement. The dependence of intensity on  $U_0$  is roughly, though not exactly,  $U_0^6$ , where  $U_0$  is the free stream speed. The turbulence length scales on the flap surface as determined by the characteristic time of the measured correlation function and the free stream speed, are from a few to many centimeters, on the order of the flap thicknesses. Time delays from the correlation between the far field signal and the surface source are determined from the correlation functions and are in good agreement with the flow-refraction-corrected results.

10:05

**KK5. Assessment at full scale of exhaust nozzle-to-wing size on STOL—OTW acoustic characteristics.** U. von Glahn and D. Groesbeck (NASA Lewis Research Center, Cleveland, OH 44135)

On the basis of static aero/acoustic data obtained at model scale, the effect of exhaust nozzle size on flyover noise is evaluated for different STOL—OTW nozzle configurations. Three types of nozzles are evaluated: a circular/deflector nozzle mounted above the wing, a slot/deflector nozzle mounted on the wing, and a slot nozzle mounted on the wing. The nozzle exhaust plane location, measured from the wing leading edge was varied from 10% to 46% of the wing chord (flaps retracted). Flap angles of 20° (takeoff) and 60° (approach) are included in the study. Initially, perceived noise levels (PNL) are calculated as a function of flyover distance at 152 m altitude.

From these plots static EPNL values, defined as flyover relative noise levels, then are obtained as functions of nozzle size for equal aerodynamic performance (lift and thrust). On the basis of these calculations, the acoustic benefits attributable to nozzle size relative to a given wing chord size are assessed.

10:20

**KK6. Intermediate range explosion airblast propagation measurements.** J. W. Reed (Environmental Research Division, Sandia Laboratories, Albuquerque, NM 87185)

Several hundred explosions of TNT, ranging in weight from 23 to 1134 kg, were fired at the NASA Space Center, Florida. Comprehensive meteorological measurements were made by rawinsonde balloons and on a nearby 150-m tower, including winds, turbulence, temperatures, and humidity. A cruciform array of airblast gauges was operated, with gauges at 200 m, 500 m, 1 km, 2 km, and 5 km ranges from the explosions. Airblast results will be correlated against refractive atmospheric conditions, in hopes of establishing a functional relationship between overpressure decay with distance and the sound velocity gradient with height. Preliminary results of the analyses will be presented. [Work supported by DOD, DOE, and NASA.]

10:35

**KK7. Aircraft noise propagation and reception anomalies at a major airbase.** Philip Dickinson (Bickerdike Allen Partners, 442 North Main Street, Bountiful, UT 84010 and University of Utah, Salt Lake City, UT 84112)

When a U.S. Air Force Air Installation Compatible Use Zoning (AICUZ) study was presented to the local government authorities in the area surrounding one major U.S. airbase, there was some disbelief and much concern about the high noise exposures predicted. So the local government authorities commissioned their own land-use compatibility study to be based on measured noise levels over an 18-month period. The results of the study only exacerbated the worries—noise levels and exposures well in excess of those predicted and large numbers of people exposed to noise more than 20 dB in excess of that generally considered "intolerable." Yet, complaints are so few as to be almost negligible, and the people seem remarkably healthy. Some of the reasons for the higher noise exposures relate to peculiar temperature effects in the high desert valley in which the base is situated. The low level of complaints is attributed to the scheduling and nature of the flights on the one hand and to the way of life of the local people on the other hand. It is suggested that the OSHA type regulations and EPA/US Air Force recommendations give a false impression of the compatibility of the environment in some circumstances.

10:50

**KK8. Acoustic wave generation by vortex shedding.** C. M. Bowline and W. Y. Abbott (215 East Truman Avenue, Apt. A, Salt Lake City, UT 84115)

It has been theorized that thrust perturbations observed in solid propellant rocket motors are caused in part by vortex shedding off flow obstructions (inhibitors), and the acoustic instability resulting from vortex flow between pairs of inhibitors. Acoustic measurements were made during wind tunnel tests developed to simulate flow between inhibitors. Results concerning geometric relationships and flow velocities are presented in this paper. In general, the flow between sets of baffles in a wind tunnel can create extremely large noises at predictable frequencies, given particular geometries and flow velocities.

11:05

**KK9. Determination of two-stroke engine exhaust noise by the method of characteristics.** Adrian D. Jones (Hills Industries Ltd., P.O. Box 78, Clarence Gardens, South Australia, 5039)

A unique computational technique was developed for the method of characteristics solution of a one-dimensional flow in a duct as applied to the wave action in an engine exhaust system. By using the method it was possible to compute the detailed unsteady flow in both straight pipe and tuned expansion chamber exhaust systems as matched to the flow from the cylinder of a small two-stroke engine. The radiated exhaust noise was then determined by assuming monopole radiation from the tailpipe outlet. Very good agreement with experiment on an operating engine has been achieved in the calculation of both the third-octave radiated noise and the associated pressure cycles at several locations in the different exhaust systems. Of particular interest is the significance of nonlinear behavior which results in wave steepening and shock wave formation. The calculation method developed differs from those of others, principally that of Blair and that of Karnopp, Dwyer, and Margolis. The method computes the precise paths on the  $x-t$  plane of a finite number of  $C_+$ ,  $C_-$ , and  $P$  characteristics, thereby obtaining high accuracy in determining the tailpipe outlet velocity and hence radiated noise.

11:20

**KK10. Application of partial coherence methods for identification of interior noise sources in aircraft.** James T. Howlett (Noise Effects Branch, NASA Langley Research Center, Hampton, VA 23665)

Effective interior noise control procedures require identification of the noise sources and the noise transmission paths. Recent developments in computational procedures have led to increased interest in partial coherence analyses for source/path determination. The theoretical aspects of this method have been explored by Bendat and also by Dodds and Robson; however, only a few reports of applications to practical problems are available. Previous authors reported only partial success in identifying sources. Therefore, improvements of the approach are felt to be needed. This paper describes the latest results of an ongoing effort to develop partial coherence techniques for interior noise source/path determination in aircraft. The paper includes a summary of the computational techniques as developed by Bendat [J. Sound Vib. **49**(3), 293–308 (1976)] and illustrates the application to a two-input, single-output system with coherence between the inputs. Solution of this system by previously used methods illustrates the occurrence of erroneous results (including negative psd) in the course of the calculation. New results are presented to suggest possible sources of the negative psd, and to suggest a procedure for avoiding this problem. The augmentation of the calculations on a digital computer interfaced with a two-channel real-time analyzer is also discussed. Future plans in the development of the partial coherence method includes application to experimental source identification in laboratory tests using simulated aircraft noise sources and fuselage transmission paths.

11:35

**KK11. Abstract withdrawn.**

11:50

**KK12. The influence of solid bodies on low mach number vortex sound.** F. Obermeier (California Institute of Technology, Applied Mathematics, Pasadena, CA 91125)

The mechanisms of sound generation by unsteady, subsonic flows in the presence of solid boundaries are investigated. For that purpose an alternative integral representation for the radiated pressure field is applied which is different from the generally used integral representation introduced by Lighthill and Curle. The main advantage of our method consists in a linear dependence of its integrand on the time derivative of the vorticity fluctuations in the hydrodynamic near field, while the ordinary Green's function has to be substituted by a "vector Green's function." This vector Green's function can be chosen for the flow fields appropriate in such a way that surface integrals do not appear. In particular the paper is concerned with two-dimensional flow and sound fields caused by a pair of spinning vortices and superimposed stationary potential flows along a finite plate or around a cylinder. Analytical solutions are determined by applying the method of matched asymptotic expansions.

FRIDAY MORNING, 30 NOVEMBER 1979

BONNEVILLE ROOM 3, 9:00 TO 12:05 P.M.

### Session LL. Physical Acoustics VII: Scattering and Related Phenomena

Steven F. Clifford, Chairman

*NOAA Wave Propagation Laboratory, Boulder, Colorado 80302*

Chairman's Introduction—9:00

#### *Contributed Papers*

9:05

**LL1. The resonances of a finite-length fluid cylinder and their interpretation in terms of surface waves.** G. C. Gaunaurd, E. Tanglis, and H. Überall (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20910)

We consider a finite-length fluid cylinder surrounded by either a very tenuous or a very dense exterior fluid. This amounts to respectively considering either Dirichlet or Neumann boundary conditions on the cylinder's surface. The eigenfrequencies of such finite-length cylinders are interpreted in this paper as the resonances caused by phase matching of circumferential waves that circumnavigate the cylinder along certain helicoidal Fermat paths, and that get reflected back and forth from its top and bottom flat surfaces. We have obtained the dispersion curves of these circumferential (i.e., "creeping") waves which are seen to correspond to a series of well-defined pitch angles of their helix, for different values of the cylinder's length-to-radius ratio. We generate and display graphs pertaining to various cylinder sizes, boundary conditions, and pitch angle of each of the resulting surface waves. [H. Überall is also at Catholic University, Washington, DC, and was additionally supported by Code 421 of ONR.]

9:20

**LL2. Critical angle diffraction in high frequency scattering by fluid spheres and cylinders.** P. L. Marston (Department of Physics, Washington State University, Pullman, WA 99164)

When plane waves are reflected from a flat surface between ideal fluids, the reflection is total when the angle of incidence  $\theta$  exceeds  $\theta_c = \sin^{-1}(c_1/c_2)$  where  $c_2 > c_1$ ;  $c_1$  and  $c_2$  are the sound speeds, respectively, of the fluid from which the wave is incident and of the second fluid. For an interface of constant curvature, ray acoustics predicts that the scattered intensity has an unphysically divergent

angular derivative as the scattering angle  $\phi$  approaches the critical scattering angle  $\phi_c = \pi - 2\theta_c$ . This divergence is removed by diffraction which is important in an angular region near  $\phi_c$ . The width of this region exceeds  $(a/\lambda)^{1/2}$ , where  $a$  is the radius of curvature and  $\lambda$  is the wavelength. A simplified approximation for the diffraction, similar to the optical analog described in "Critical angle scattering by a bubble: physical-optics approximation and observations" [P. L. Marston, J. Opt. Soc. Am. **69**, 1205-1211 (1979)], is derived. As  $\phi$  approaches  $\phi_c$ , a ringing and decay of the far-field intensity is predicted which may be observable in scattering by fluid spheres and cylinders.

9:35

**LL3. Ultrasonic nonspecular reflectivity near longitudinal critical angle.** T. D. K. Ngoc and W. G. Mayer (Physics Department, Georgetown University, Washington, DC 20057)

The intensity profile of an ultrasonic beam reflected from a liquid-solid interface is determined by a numerical integration method. This numerical approach takes into account the influence of absorption in the media and is valid for all angles of incidence. The reflected profile is calculated for a specific case: incidence near the longitudinal critical angle for a water-Plexiglas interface. The calculated results demonstrate the existence of nonspecular reflectivity near this particular critical angle and provide a quantitative description of its basic features. Theoretical results and experimental measurements are compared. [Work supported by the Office of Naval Research, U.S. Navy.]

9:50

**LL4. Scattering of sound from a randomly rough solid-liquid interface.** S. K. Numrich (Naval Research Laboratory, Washington, DC 20375)

Measurements have been made of sound scattered by submerged surfaces with Gaussian height and slope statistics. A wide range of roughness was spanned through the use of transient pulse analysis. In terms of the ratio of the rms surface height ( $\sigma$ ) to sound wavelength ( $\lambda$ ), the range included  $0.03 < (\sigma/\lambda) < 5.0$ . Coherent and incoherent terms were separated digitally. In the region of small roughness, the coherent reflection coefficient was compared with two theoretical models. Comparisons with theory were also made using the incoherent term when large values of roughness were involved. At both extremes of the range, the experimental data agreed well with the theoretical predictions. Of particular interest in this experiment is the intermediate region in which the coherent and incoherent terms are explained in detail. Here the coherent term decreases rapidly and the incoherent term begins to dominate the scattering return. No adequate theoretical model exists for this transition region. [Work supported by ONR, Code 461, and NOSC, San Diego.]

10:05

**LL5. Backscattering of short ultrasonic pulses by solid elastic cylinders at large  $ka$ .** P. J. Welton (Westinghouse Electric Corporation, P.O. Box 1488, Annapolis, MD 21404), M. De Billy, A. Hayman, and G. Quentin (Groupe de Physique des Solides, de l'Ecole Normale Supérieure, Université Paris VII, Tour 23-2, place Jussieu, 75221 Paris Cédex 05, France)

Backscattering measurements from solid brass, aluminum, and Lucite cylinders in water have been performed at a  $ka$  of approximately 240 using short acoustic pulses. Numerous discrete echoes are received due to multiple internal reflections of the acoustic pulse at the boundary of the cylinders. Excellent agreement between the measured echo arrival times and the echo arrival times predicted by the theory of Brill and Überall [D. Brill and H. Überall, J. Acoust. Soc. Am. **50**, 921-939 (1971)] is obtained for all of the cylinders. The amplitudes of the backscattered echoes from the brass and aluminum cylinders were also compared with theory and fair agreement is obtained.

10:20

**LL6. The scattering of an obliquely incident plane acoustic wave from a cylindrical object.** Lawrence Flax (Naval Research Laboratory, Washington, DC 20375) and Vasundara V. Varadan, and Vijay K. Varadan (Wave Propagation Group, Department of Engineering Mechanics, The Ohio State University, Columbus, OH 43210)

A mathematical model is developed to predict the scattering of a plane acoustic wave incident upon an elastic cylinder immersed in water at any angle relative to the cylindrical axis. Solutions to the elastic problem are constructed using scalar and vector potentials. There are three important angular regions of interest. These are at the longitudinal, shear, and Rayleigh critical angles. The scattering amplitude at these angles is derived and will be discussed.

10:35

**LL7. The  $T$ -matrix approach to scattering of waves by finite elastic and viscoelastic cylinders immersed in water.** Vasundara V. Varadan, Vijay K. Varadan (Wave Propagation Group, Department of Engineering Mechanics, The Ohio State University, Columbus, OH 43210), and Lawrence Flax (Naval Research Laboratory, Washington, DC 20375)

In this paper, the  $T$ -matrix or null field approach is applied to make scattering and absorption calculations for finite elastic and viscoelastic cylinders in water. Various difficulties which so far have prevented the use of such a formalism for a solid-fluid interface will be discussed. The complication due to the appearance of only curl free basis functions in water which in turn makes some of the

matrices involved nonsquare has been overcome by considering additional representation of the scattered and refracted fields and using a series of matrix manipulations. Numerical results displaying the scattering cross sections for a range of frequencies for various aspect ratios are presented. These results are then compared with those of a prolate spheroid of the same overall dimensions to study how the shape is critical in determining the absorption characteristics.

10:50

**LL8. "Zeroes," "ridges," and poles, in the scattering amplitudes of elastic waves echoing from resonating fluid spheres in solids and liquids.** D. Brill (U.S. Naval Academy, Physics Department, Annapolis, MD 21402), G. Gaunard, and H. Überall (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20910)

We study the "resonance parts" of the scattering amplitudes of compressional or shear waves, returned when compressional or shear waves are incident on fluid spheres contained in either solids or in dissimilar fluids. The resonance parts are obtained after suitable smooth backgrounds are subtracted from each composite partial-wave amplitude. The starting point of the analysis is the three-dimensional graph of any of the modular surfaces  $|{}_a f_n^{pp}|$ ,  $|{}_a f_n^{sw}|$ ,  $|{}_a f_n^{sp}|$ , and  $|{}_a f_n^{ss}|$  plotted versus mode order  $n$ , and nondimensional frequency  $ka$ . We found simple expressions for the (not necessarily straight) lines of zeros that constitute the boundaries or demarcation lines of each of the "ridges" that appear so evidently in the plots. These "ridges," which are inclined with respect to the  $n$  and  $ka$  axes, exhibit marked cleavages, and the "peaks" present in them (i.e., the resonances) are spaced in a peculiar way that is, (a) predictable from the analysis, and (b) that yields information about the material composition of the scatterer. The analytic expressions for the lines connecting the tips of the peaks along a given ridge has also been determined. Each "ridge" is shown to be associated with a "creeping wave" and it is the graphic visualization of the elastic analog of a "Regge pole." [H. Überall is also at Catholic University, Washington, DC, and was additionally supported by Code 421 of ONR.]

11:05

**LL9. The Mittag-Leffler expansion and the interpretation of poles in the scattering amplitudes of waves echoed from fluid spheres.** G. Gaunard and H. Überall (Naval Surface Weapons Center, White Oak, Silver Spring, MD 20910)

We consider the cases of acoustic and elastic wave scattering from fluid (possibly viscous) spheres contained within either solids (i.e., coatings) or dissimilar fluids (i.e., bubbles), and study the resulting scattering matrix  $S_n$  and the poles in the associated scattering amplitude ( $s$ ). The one-term Taylor series we used in the past to obtain the "one-level approximation" to  $S_n$ , which was so valuable in our earlier development of the resonance theory of viscoelastic wave scattering from cavities, can now be replaced by the exact Mittag-Leffler expansion of the quantity  $F_n$ . This quantity is proportional to the system's mechanical impedance and since all of its singularities are simple poles, it admits a meromorphic function (i.e., Mittag-Leffler) series representation. The locations of the "new" resonances and the value of their widths are found to be shifted from the values we found using our earlier approach, even in the absence of absorption in the fluid interior to the sphere. Absorption introduces further shifts and the unitarity property of  $S_n$  is lost in both instances. The new representation is not only exact, but reduces properly to the right asymptotic limits and our earlier "backgrounds" and "resonances" can be re-interpreted in the light of these new (and simpler to obtain) poles. Plots of the poles and their shifts are displayed in various cases. [H. Überall is also at Catholic University, Washington, DC, and was additionally supported by Code 421 of ONR.]

**LL10. A simple scattering model for finite objects.** J. C. Nelander and A. D. Matthews (Naval Coastal Systems Center, Panama City, FL 32407)

A model for acoustic scattering from finite objects is described in terms of a three-dimensional Fourier wave space. This wave space representation, adapted from x-ray diffraction analysis, provides a convenient method for depicting aspect and frequency dependence simultaneously. Physical observables are interpreted by means of the Ewald construction. The method is illustrated for a rigid right circular cylinder. A procedure which eliminates mathematical difficulties which arise from surface discontinuities for finite objects when one-dimensional models are employed is discussed. Incorporation of impedance boundary conditions can alter details of the cylinder's scattering cross section. Examples are discussed and illustrated for aluminum and brass cylinders in water. Dominant observed features are related to the large phase shift which occurs at the shear critical angle. The model provides a useful simulation tool and also suggests a method for analyzing experimental data in detail.

**LL11. Creeping wave description of surface wave modes on elastic cylindrical shells.** J. W. Dickey, D. A. Nixon (David W. Taylor Naval Ship Research and Development Center, Annapolis, MD 21402), and E. D. Breitenbach (David W. Taylor Naval Ship Research and Development Center, Bethesda, MD 20084)

Dispersion and attenuation curves are given for identified surface modes in the scattered pressure from elastic cylindrical shells insonified by a normally incident plane wave. The results are derived using two different techniques for air-filled aluminum shells of several different wall thicknesses,  $b/a$  (ratio of inner to outer radius). The first series of numerical results are calculated using the Sommerfeld-Watson transformation and determining the roots of a  $6 \times 6$  secular determinant. The second method is used to derive similar results

by analysis of the partial wave responses. The second method results, for a thick shell, are favorably compared with previously obtained creeping wave results for a solid elastic cylinder. For low values of  $b/a$ , the Rayleigh and Stoneley waves on the shell are shown to approach limiting values for increasing  $ka$  which correspond to the Rayleigh and Stoneley wave speeds defined for the infinite half-space, and the Franz and Whispering Gallery modes approach the bulk wave velocities for the external and shell materials respectively. For thin shells ( $b/a \rightarrow 1$ ), Whispering Gallery and Rayleigh modes have vanished and the symmetric and antisymmetric Lamb modes are prevalent over limited regions of  $ka$ . Calculations are carried out over the range  $0.1 \leq ka \leq 200$ , the first method valid for higher  $ka$ , and the partial wave method valid at lower ranges of  $ka$ .

**LL12. Scattering of elastic waves by an elliptical cylindrical shell.** M. M. Simon (Analysis and Technology Inc., North Stonington, CT 06359) and R. P. Radlinski (Naval Underwater Systems Center, New London Laboratory, New London, CT 06320)

Much work has been accomplished in the analysis of the dynamics of an elliptical cylindrical shell. Recently, solutions have appeared for acoustic scattering from a submerged elliptical shell [Brigham, Libuha, and Radlinski, *J. Acoust. Soc. Am.* **61**, 48-59 (1977)]. The problem of scattering of arbitrarily polarized elastic waves from an elliptical cylindrical shell is even more difficult because the basis functions for the fields in a separable coordinate system are not orthogonal for the different wave speeds. Recent advances [Pao and Varatharajulu (Varadan) *J. Acoust. Soc. Am.* **59**, 1361-1371 (1976)], in the application of the extended boundary condition method to elastic wave propagation has led to solutions for elastic wave scattering from elliptical inclusions that are stress-free, rigid, fluid, or solid elastic. Varadan and Varadan have worked on the case of a layered elastic inclusion. This presentation deals with coupling of thin shell theory to the elastic scattering problem as described by the extended boundary condition method. The theory is ideally suited to such applications since the formulation is given in terms of normal and tangential stresses and displacements on the elliptical surface.

FRIDAY MORNING, 30 NOVEMBER 1979 GRAND BALLROOM I, 9:00 A.M. TO 12:00 NOON

## Session MM. Psychological Acoustics VI: Binaural Hearing and the Perception of Complex Tones and Patterns (Poster Session)

Walt Jesteadt, Chairman

*Boys Town Institute for Communication Disorders in Children, Omaha, Nebraska 68131*

### Contributed Papers

**MM1. The Shepard demonstration revisited, again.** E. M. Burns (Department of Audiology and Speech Sciences, Purdue University, West Lafayette, IN 47907)

The Shepard illusion ("auditory stairway to heaven"), in which the presentation of a cyclically repetitive sequence of complex tones composed of partials separated by octave intervals gives the illusion of continually rising pitch [Shepard, *J. Acoust. Soc. Am.* **36**, 2346-2353 (1964)] is often cited as evidence for octave equivalence in relative pitch judgments. Evidence will be presented which demonstrates that this illusion can be produced using inharmonic complex tones (e.g., tones whose partials are separated by stretched or compressed "octaves") and, thus, octave separation

of partials is not a necessary condition for producing the illusion. [Work supported, in part, by NINCDS.]

**MM2. Discrimination between random signals.** Irwin Pollack (Mental Health Research Institute, University of Michigan, Ann Arbor, MI 48109)

Random polarity pulse trains of two states (plus and minus polarity) and of three states (plus and minus polarity and no change at defined clock intervals) were constructed and presented to listeners within forced-choice discrimination tests. The pulse trains were either of the same average pulse density or were amplitude-

compensated for different pulse densities. Since the long-term spectrum of such signals is uniform, the most likely discriminable feature is the short-term spectrum or the temporal patterning of the signals. Successful discrimination is obtained even for pulse trains of extremely high clock frequencies. [Work supported by NSF.]

**MM3. The relation between gap discrimination and auditory stream segregation.** D. L. Neff, W. Jesteadt (Boys Town Institute for Communication Disorders in Children, Omaha, NE 68131), and E. Brown (Department of Psychology, University of Nebraska at Omaha, Omaha, NE 68182)

The relation between gap discrimination and stream segregation [A. S. Bregman and I. Campbell, *J. Exp. Psychol.* **89**, 244–249 (1971)] was examined using sequences of four 200-ms, eight 100-ms, or sixteen 50-ms tones with a total frequency range from  $\frac{1}{4}$  to  $2\frac{1}{2}$  octaves. A gap of variable duration was inserted between tones of similar or disparate frequency. Gap discrimination, in a 2IFC procedure, became poorer as the difference between the frequencies of the tones bounding the gap increased, but was not influenced by changes in sequence length and rate. Judgments of stream segregation for the same stimuli by the same listeners showed strong individual differences, an influence of sequence length and rate as well as marker-frequency difference, but little relation to gap discrimination. [Work supported by Grant NS-14709 from NINCDS.]

**MM4. Temporal asynchrony within auditory patterns.** J. D. Gilliom and S. L. Brady (Department of Psychology, University of Alabama in Birmingham, Birmingham, AL 35294)

Another in a series, the present study explores the effect of temporal asynchrony on the perception of part tones within the context of a musical chord. The basic paradigm requires an observer to perform a monaural yes–no detection task of a tonal signal within Gaussian noise, with signal frequency a random variable. A contralateral frequency cue is provided on each trial prior to the detection interval, enabling an observer to reduce the negative effects on detection performance of such signal frequency uncertainty. The experimental manipulation consists of placing this cue within the context of a three-element musical chord and varying the onset and offset of the cue component relative to the other components of the chord while monitoring the cue's effectiveness in raising detection performance. Shifts in detection performance from such manipulations indicate that the availability of specific spectral pitch information from a part tone within a musical chord is substantially enhanced by making its onset and/or offset asynchronous with that of the remaining components of the musical chord.

**MM5. Temporal and spectral determinants of informational masking in tonal patterns.** Charles S. Watson, William J. Kelly, and Douglas M. Mellen (Boys Town Institute for Communication Disorders in Children, Omaha, NE 68131)

A form of "informational masking" is observed when listeners attempt to detect single target components of word-length tonal patterns [*J. Acoust. Soc. Am. Suppl.* **1** **64**, S39 (1978)]. These masking-like effects, sometimes as large as 50–60 dB, occur under conditions of high stimulus uncertainty, when contextual patterns and the temporal and spectral positions of target tones are varied from trial-to-trial, but are absent under conditions of minimal psychophysical uncertainty. Stimulus determinants of informational masking were investigated in three experiments, by (a) varying the duration of the contextual patterns from 0 to 400 ms, (b) varying the duration of the target tones from 20 to 160 ms, and (c) varying the spectral range for the contextual tones from 1 to 2700 Hz. Individual differences in asymptotic detection thresholds, and in training times required to approach those thresholds, are unusually large in these tasks. However, the data are generally characterized by the following rules. Informational masking approaches asymptotic

levels for context durations in excess of 160 ms. Little or no masking occurs for target tones longer than 60–80 ms. Similar average amounts of masking were observed for each of the spectral ranges of the context tones. [Work supported by NIH.]

**MM6. Effects of neuropharmacological agents on auditory processing skills in young adults.** C. Ludlow, E. Caine, E. Cudahy, and C. Cardano (NINCDS, NIH, Federal Building, Room 1C-41, Bethesda, MD 20205)

Two commonly used compounds (scopolamine and dextro-amphetamine) well known for acutely manipulating brain function were administered to normal young adults 1 h prior to performing auditory tasks in a double blind placebo-drug crossover study. A gap detection task required the detection of brief periods of silence in noise using a forced choice procedure. A temporal order task required identification of order of a two tone sequence using a forced-choice procedure. Thresholds for both were derived from least square fitted psychometric functions. Scopolamine, an acetylcholine inhibitor, significantly increased the gap size detected at 75% threshold and increased the ISI at 75% threshold on the temporal order task. The effects of dextroamphetamine, which augments catecholamine, mediated neurotransmission of brain arousal were much less dramatic. No change occurred on the gap detection functions while a very small effect towards improving the ISIs occurred on the temporal order task. Thus, increased brain arousal did not improve auditory processing skills while reduction in temporal lobe acetylcholine content adversely affected the processing of rapidly changing acoustic events.

**MM7. The precedence effect revisited.** C. M. Brandauer (Bell Laboratories, 11900 North Pecos, Denver, CO 80234) and W. H. Ward (Psychology Department, University of Colorado, Boulder, CO 80309)

In their now classical study of the precedence effect, Wallach, Newman, and Rosenzweig [*Am. J. Psychol.* **62**, 315–336 (1949)] determined the interaural time difference of the first click pair (ITD1) needed to offset a fixed and opposite interaural time difference of a second click pair (ITD2) when the pairs were separated by a time interval short enough to yield a single, fused image. For several values of ITD2, they found that the indifference point, i.e., where "right" and "left" judgements were equally likely, occurred when  $|ITD2|/|ITD1| = 6$ . We have repeated their experiment at various sensation levels and found that the  $|ITD2|/|ITD1|$  ratio decreases from values greater than theirs at SL = 20 dB to nearly one at SL = 65 dB. The relationship of these results to earlier studies and some possible explanations will be discussed.

**MM8. Diotic and dichotic perception of single echoes.** Arthur H. Koenig (Bell Laboratories, Holmdel, NJ 07733 and Communication Science Laboratory, City University of New York, New York, NY 10036)

The perception of single echoes for two binaural listening modes, diotic and dichotic was examined. Stimuli consisted of a white Gaussian noise waveform added to itself with an intervening delay. Delay values of 2, 4, 8, and 16 ms at two overall sensation levels of 30 and 50 dB SL were used. The dichotic listening mode was created by reversing the polarity of the delayed noise waveform in one ear relative to the other ear. An ABX paradigm was used to determine the absolute threshold for the presence of a single echo. Diotic listening mode thresholds were found to be significantly different from dichotic listening mode thresholds. For delay values of 2 and 4 ms, the diotic listening mode thresholds were substantially elevated relative to the dichotic listening mode thresholds. Diotic threshold data appear to be compatible with existing critical-band models. Dichotic threshold results lend support to a mechanism of binaural combination which serves to reduce spectral variation.



**MM9. Handedness, sex, ear effect, and children's contralateral acoustic reflex.** D. W. Johnson (Audiology, Hennepin County Medical Center, Minneapolis, MN 55415) and R. E. Sherman (Hennepin County Government Center, Minneapolis, MN 55415)

There has been no clear evaluation of handedness and sex on the contralateral acoustic reflex with aging. To investigate these effects and their interactions, 54 children aged 6 to 12 yr were selected from normal classroom settings on the basis of age, sex, handedness for writing, average academic performance, normal hearing, and normal medical history. Contralateral acoustic reflex thresholds were determined using an ascending technique with continuous white noise stimuli. When all children were grouped together, there was no age or sex effect observed; the right ear did appear to average 3.8 dB more sensitive to white noise stimuli than the left. When grouped by handedness, left-handed youngsters showed a pronounced age effect, thresholds improving 12 dB from age 6 to 12; right ears continued to be the superior ear for these left-handed children. Left-handed children were approximately 2 years later in developing contralateral acoustic reflex sensitivity equal to that of right-handed peers, but appeared to mature faster than right-handed children from ages 8 to 12 in terms of acoustic reflex threshold sensitivity.

**MM10. Infant and adult ear asymmetry differences for memory-based consonant versus vowel discriminations.** Catherine T. Best (Haskins Laboratories, New Haven, CT 06510)

Recent evidence indicates that the adult REA for speech and LEA for music are present by 3 months. Yet the nature of perceptual asymmetries, and the effect of stimulus properties, in the infants' speech REA are unknown. The adult literature on perceptual asymmetries suggests a possible left hemisphere mechanism for perception of consonants via reference to rapid articulatory gestures, since adults typically show REAs for consonants but not for vowels. Condition A of this study measured infant and adult ear asymmetries for discriminations among consonants versus vowels in CV syllables. Studies of associations between stimulus properties and adult ear asymmetries indicate special left hemisphere responsivity to transient information, e.g., rapid formant transitions. Therefore, Condition B assessed how removal of formant transitions from the CVs (phonemes remained identifiable) affected infant and adult ear asymmetries for consonant versus vowel discriminations. Results indicate adult asymmetries are largely perceptual, with left hemisphere phonetic perception for consonants but not vowels. However, acoustic factors apparently outweigh phonetic perceptual factors in infant speech REAs. Implications for brain-behavior relationships in perceptual development, especially speech perception, will be discussed. [Partially supported by NICHD and NINCDS.]

**MM11. Brain lateralization in 2-, 3-, and 4-month olds for phonetic and musical timbre discriminations under memory load.** Catherine T. Best (Haskins Laboratories, New Haven, CT 06510), Harry Hoffman (North Dakota State University, Fargo, ND 58102), and Bradley B. Glanville (California State University, Chico, CA 91710)

Although young infants show the adult pattern of dichotic ear asymmetries, it is unclear whether any early age changes exist in the pattern or strength of infant asymmetries. Knowledge about early lateralization may illuminate issues in perceptual development. In this study, 2-, 3-, and 4-month olds completed dichotic tests for ear differences in short term memory-based discriminations among synthetic syllable-initial stop consonants, and among synthesized renditions of different instruments playing one note. Discrimination results supported an REA for speech, and an LEA for musical timbre, in 3- and 4-month olds. The 2-month olds showed only the musical timbre LEA, without reliable syllable-initial stop consonant discrimination by either hemisphere under memory load.

Of the individual infants, an LEA was found in  $\frac{3}{4}$  of those who discriminated timbre contrasts, and an REA was found in  $\frac{3}{4}$  of those who discriminated speech contrasts. Both portions are close to those found in adults. Implications for theories about lateralized brain development, and for infants' perception of the two classes of complex auditory stimuli, will be discussed. [Partially supported by NINCDS and NICHD.]

**MM12. A comparison of localization learning in a monaural aided condition in noise with trained and nontrained listeners.** J. L. Bunce, R. H. Brey, and K. O. Jones (Communicative Disorders Area, Brigham Young University, Provo, UT 84601)

A preview study was conducted to look at localization learning in monaural listeners. Eight subjects with hearing ability within normal limits were selected for the study. The subjects were asked to determine which speaker a signal was coming through with one aided and one plugged ear. The eight subjects were divided into two groups of four, an experimental and a control group. Each subject was tested in three 1 h sittings. Testing was accomplished in an anechoic chamber with eight speakers at 45° intervals. The signal was a narrow band noise, 2400–4800 Hz. Two signal-to-noise ratios were selected, –5 and –10. The noise was white noise. Subjects were allowed to move their heads 180°, 90° to either side. The experimental group received immediate feedback as to the actual speaker the signal was coming through and were allowed to take notes of any clues they could acquire. The control group received no feedback. The results indicated that learning takes place with and without training. However, those subjects who received training exhibited significantly better scores.

**MM13. Effects of musical training on ear advantage for a dichotic rhythm task.** J. D. Craig (USA Human Engineering Laboratory, Behavioral Research Directorate, Aberdeen Proving Ground, MD 21005)

Four right-handed musicians and six right-handed nonmusicians responded to four-beat dichotic rhythm patterns by using a same/different comparison method. The subjects were tested for laterality in response to the rhythm patterns at five frequencies: 400, 800, 1600, 3200, and 6400 Hz. The results indicated that there was a clear right-ear advantage in the response of nonmusicians to rhythms of 800, 1600, and 3200 Hz, and a slight left-ear advantage for 400- and 6400-Hz patterns. There was virtually no ear advantage in the response of musicians for rhythms of 400 through 3200 Hz, but a definite left-ear advantage was obtained for 6400-Hz patterns. The results support an hypothesis that musical training develops the rhythm areas in the right hemisphere which may be antithetical to the left hemispheric speech structures.

**MM14. Effects of training and head movements on binaural and monaural localization.** R. W. Gatehouse and P. J. Russell (Department of Psychology, University of Guelph, Guelph, Ontario, Canada, N1G 2W1)

In delineation of localization variables, research has focused primarily on single "main effects." Therefore, the horizontal and vertical localizing accuracy and bias of 72 normally hearing subjects was investigated under monaural versus binaural, restricted versus nonrestricted head movement, and training with or without feedback listening conditions. A counterbalanced mixed factorial design allowed the parcelling out of the significant interactive effects in a multivariate study that more closely approximates everyday free field localization. The to-be-localized signal was a 2-s burst of pulsed WN (25 dB, SL) which could come from any of 12 azimuths (0, 30°, 60°, . . . , 330°) at any one of 3 elevations (0° ± 30°). Although some well-documented effects were confirmed, e.g., binaurals localize better in both planes; sources are shifted to unplugged ear side, the results demonstrate that no simple relationships exist between

training and head movement conditions. For instance, feedback training benefits monaurals more than binaurals, while head fixation debilitates binaurals more as seen by larger errors and more back-front confusions. The results will be discussed in reference to the "variable weighting" approach of the Searle *et al.* model [J. Acoust. Soc. Am. 60, 1164-1175 (1976)] of localization.

**MM15. Phase effects in pure tone central masking.** H. B. Calder (Speech and Hearing Sciences, University of Michigan, Ann Arbor, MI 48109)

Centrally masked threshold shifts were determined for 1000-Hz pure tone signals masked by a 200-ms, 1000-Hz masker with either 0° or 180° interaural phase disparity between the signal and the masker. Three signal durations of 20, 100, and 200 ms were employed in the study. Differences in the masked thresholds of 1.5 to 5.6 dB were occasioned by the phase reversal of the signal and masker with the greatest difference occurring for the 200-ms duration signal. These results support models which account for central masking on the basis of a physiological interaction at the level of the brainstem. [Work supported by Horace Rackham Graduate School.]

**MM16. Detection of a tone below absolute or tone-masked threshold by interaural phase discrimination.** Thomas Ayres and T. D. Clack (Kresge Hearing Research Institute, Ann Arbor, MI 48109)

Binaural beating or changes in the threshold of an audible tone are sometimes reported when an inaudible tone is added to the contralateral ear. This study investigates a related effect using a two-interval forced-choice task. In this procedure, the subthreshold signal (800 Hz) is presented continuously to one ear; a probe tone (also 800 Hz, but at 10 dB SL) is presented in two pairs of short bursts (250 ms) to the contralateral ear. An interaural phase shift of 180° is introduced at the second burst of one pair, and subjects are asked to detect any change. Listeners usually hear these phase effects as variations in lateral position, even when the signal is 10 dB or more below its threshold (4 out of 6 subjects). One achieved above-chance performance with the signal at -15 dB SPL (-21 dB SL). Such results show that information regarding a subliminal monaural tonal signal may be transduced and available to the binaural-processing system even at such extremely low sound pressure levels. Similar results are found when the threshold is shifted by using a continuous tone as a masker (300 or 500 Hz at 60-75 dB SPL). These findings are strikingly different from those obtained with the masking-level difference paradigm.

**MM17. Binaural interaction with an aural harmonic: Cancellation and augmentation.** Thomas Ayres and T. D. Clack (Kresge Hearing Research Institute, Ann Arbor, MI 48109)

In the monaural tone-on-tone masking procedure, the audibility of a maskee an octave above the masker depends on masker:maskee phase. If vector summation between the maskee and an aural harmonic is assumed, the amplitude and phase of the aural harmonic can be estimated. An alternative interpretation of the phase effects is two-tone interaction between the masker and maskee. To decide between these explanations, a binaural technique has been developed for the study of octave phase effects. Using a forced-choice procedure, listeners are able to detect a change in interaural phase between a 400-Hz continuous masker presented to one ear (65-75 dB SPL) and an 800-Hz pulsed probe tone presented contralaterally (10 dB SL). When a subthreshold 800-Hz tone is added to the masker ear at a phase and level appropriate to cancel the estimated second aural harmonic, performance deteriorates significantly. Addition of this same tone at augmentation phase tends to improve performance. These findings fail to confirm the two-tone interaction interpretation, but they support the aural-harmonic explanation for monaural tone-on-tone masking results. Furthermore, these inaudible "subjective" products must be represented in the neural input to the brainstem

since, like inaudible acoustic tones, they are capable of interacting binaurally with audible tones.

**MM18. Short duration adaptation effects upon lateralization.** Donald N. Elliott and Changiz Geula (Department of Psychology, Wayne State University, Detroit, MI 48202)

Changes in interaural phase relations or intensity differences required to return a binaural intercranial image to the median plane following adaptation of one ear have been determined for three Os. The monaural adapting signal was 500 Hz at 60 dB SL, and of 150-, 300-, or 1000-ms duration. The 300-ms binaural signal input to be judged for location was also 500 Hz, and was varied either in interaural phase or intensity in order to produce a median plane lateralization (MPL). This binaural signal followed the offset of the adapting signal by 0, 30, 1000, 2000, 5000, or 10 000 ms. It was found that sizable interaural phase or intensity differences are required to produce a median plane image following adaptation. The effect develops very rapidly and lasts for an extended period—for some Os being evident 10 s after a 150-ms adapting signal. Time/intensity ratios required for MPL were found to increase with increasing adaptation effects. Sizable asymmetries were found for some Os, with far larger effects evident after adapting one ear than the other.

**MM19. Release from masking for speech under pseudodichotic listening conditions.** G. C. Tolhurst (Communication Disorders Department, University of Massachusetts, Amherst, MA 01003)

A series of studies has shown that scores of a multiple-choice speech reception test are significantly higher if the signals to one ear are processed by certain bandpass filtering or are interrupted at certain rates while the same signal to the opposite ear remains as undistorted as possible, i.e., pseudodichotic, over the condition of undistorted speech to both ears. No release from the white noise masking was found when the speech signals to one ear were time delayed variously to that presented to the opposite ear. The above results pertain whether the white noise was mixed with speech electronically or mixed acoustically. The release from masking was more pronounced at a plus 6 dB S/N than at a plus 2 dB S/N. Different groups of 30 to 35 normally hearing listeners made responses to 12 different randomizations of the test, one each for, (a) delay, interruption, or filtering signal processing (3), (b) electronic or acoustic noise mixing (2), and plus 6 dB or plus 2 dB S/N conditions (2).

**MM20. Vertical sound localization in monkeys.** Charles H. Brown (Department of Psychology, University of Missouri, Columbia, MO 65211)

Minimum audible angles for vertical localization were psychophysically determined in Old World monkeys (*Macaca*). Monkeys were trained through positive reinforcement operant conditioning procedures to contact a response disk (observing response) initiating a repetitive series of brief pulses (approximately 300 ms) presented from a standard location directly in front of the monkey at 0° elevation. At random intervals the stimulus changed location from the standard to one of four comparison locations. The monkeys reported the change in elevation by releasing the response disk. The minimum audible angle for vertical localization was assessed through the method of constant stimuli under free-field conditions in an anechoic chamber. The test stimuli consisted of primate vocalizations and bands of noise adjusted in bandwidth. Minimum audible angles ranged from approximately 3° to 20° for signals of various bandwidths. Monkeys were unable to detect changes in elevation for signals with upper cutoff values (of the spectrum) below 2000 Hz. The results suggest that minimum audible angles for vertical localization were dependent upon torso reflections rather than pinna transformations. [Work supported by NSF.]

Session NN. Speech Communication VII: Vowel and Consonant Perception (Poster Discussion Session)

Dennis H. Klatt, Chairman

*Research Laboratory of Electronics, Massachusetts Institute of Technology,  
Cambridge, Massachusetts 02139*

Chairman's Introduction—9:00

Contributors will be at their posters from 9:00–10:30 A.M. The chairman will lead a discussion of pertinent issues from 10:30 A.M. to 12 NOON.

*Contributed Papers*

**NN1. Perceptual comparisons among a set of vowels similar to /æ/:** Some differences between psychophysical distance and phonetic distance. Dennis H. Klatt (Room 36-523, Massachusetts Institute of Technology, Cambridge, MA 02139)

A set of 66 vowels acoustically similar to /æ/ were synthesized by adding together sinusoidal harmonics of the appropriate frequencies, amplitudes, and phases [Carlson, Granström, and Klatt, *J. Acoust. Soc. Am. Suppl. 1* **65**, S6 (1979)]. Subjects were asked to estimate either (1) the psychophysical distance, or (2) the phonetic distance between each stimulus and a reference vowel in a 300-trial randomized test. Stimulus manipulations included changes to formant frequencies, formant bandwidths, spectral tilt, phase relations among harmonics, vocal tract length, and filtering passband/stopband. Results indicate that when subjects are making phonetic comparisons among the stimuli, they pay far less attention to changes in phase, filtering passband, spectral tilt, and formant bandwidths than when making psychophysical judgements. An acoustically based distance metric consistent with the phonetic data is being sought. [Work supported by an NIH grant.]

**NN2. On the identification of synthesized steady-state vowels in isolation and in consonantal context.** Randy L. Diehl, Susan Buchwald McCusker, and Laura S. Chapman (Department of Psychology, University of Texas at Austin, Austin, TX 78712)

Naturally produced vowels have been shown to be much more difficult to identify correctly in isolation than in consonantal context [W. Strange, R. R. Verbrugge, D. P. Shankweiler, and T. R. Edman, *J. Acoust. Soc. Am.* **60**, 213–224 (1976)]. In an attempt to replicate this finding under more controlled conditions, we synthesized three sets of English monophthongs. The first set consisted of 240-ms steady-state formant patterns bounded by symmetrical transitions corresponding to a b\_b context; the second consisted of the 240-ms steady-state segments in isolation; and the third set was identical to the second, except that the vowels were lengthened to equal the duration of the CVC syllables of set one. Surprisingly, both the short and long isolated vowels were identified with greater accuracy than the consonant-bounded vowels. In a second experiment, we sought to make the stimulus sets more natural by (1) introducing duration differences among vowels within each set corresponding to differences observed among naturally produced vowels and (2) reducing the transition extent of the CVC stimuli from 500 to 250 Hz. Overall error rates were substantially reduced. The two sets of isolated vowels were still identified slightly more accurately than the consonant bounded vowels, but the difference was no longer significant. [Work supported by NINCDS.]

**NN3. Acoustic cues in the identification of vowel sequences.** J. E. Kerivan (NSMRL, Auditory Department, Submarine Base, Groton

CT 06340) and Peter J. Alfonso (Department of Speech, University of Connecticut, Storrs, CT 06268)

We have observed that at 30- and 60-ms durations some permuted orders of 4 synthetic vowels /i, æ, a, u/ were much more identifiable than others using a single sequence method [Kerivan, Alfonso, and Bornstein, *J. Acoust. Soc. Am. Suppl. 1* **65**, S113(A) (1979)]. Also, significant improvements in identification of difficult sequences occurred when silent gaps were placed between selected vowels resulting in a rhythmic grouping of the sequence. These findings have been correlated to the various acoustic parameters used to synthesize the vowels. The results of these multiple correlations show that the identification of vowels in sequence follows the laws of pitch continuity established by numerous investigators using tonal sequences. The more monotonic pitch continuity displayed by adjacent first formants the better is sequence identification. A similar but weaker correlation was found between sequence identification and pitch proximity of fundamental frequencies. An interaction was also found between pitch and loudness of adjacent fundamental and formant frequencies. The relative intensity differences for all first formants were within 3 dB; however, those for fundamental frequencies were as disparate as 11 dB. It appears that the ear primarily tracks the first formant of each vowel complex in determining vowel identity in the sequence. [Work supported by an NINCDS NIH grant awarded to Haskins Laboratories and the University of Connecticut Research Foundation.]

**NN4. Rhythmic grouping effects on temporal order identification of vowel sequences.** Peter J. Alfonso (University of Connecticut, Storrs, CT 06268) and John E. Kerivan (NSMRL, Auditory Department, Submarine Base, Groton, CT 06340)

Near-perfect performance has been shown for identification of four synthetic vowel sequences /i, æ, a, u/ at 90–150-ms vowel durations. However, a significant permuted order effect was found for certain 30- and 60-ms vowel sequences by Kerivan, Alfonso, and Bornstein [*J. Acoust. Soc. Am. Suppl. 1* **65**, S113(A) (1979)]. In a further experiment, we have investigated rhythmic grouping effects on the 30- and 60-ms vowel sequences which displayed the permuted order effect in the first experiment. Rhythmic grouping was created by (1) the introduction of a silent gap between two vowels in a sequence, or (2) varying the duration of a single vowel within a sequence. All possible permutations of the two rhythmic grouping conditions were created on the Haskins serial resonance synthesizer by modifying the original stimuli. The results of the experiment indicate that the permuted order effect can be significantly reduced in steady-state concatenated vowel sequences by controlling certain prosodic temporal features of the sequences. [Work supported by an NINCDS NIH grant awarded to Haskins Laboratories and the University of Connecticut Research Foundation.]

**NN5. The role of formant transition duration in the perception of stress.** L. J. Raphael (Lehman College, CUNY, Bronx, NY 10468) and Haskins Laboratories, New Haven, CT 06511) and M. F. Dorman (Arizona State University, Tempe, AZ 85281)

The results of experiments previously reported indicated that syllable-initial formant transitions contribute by their duration to the perception of the syllable-final voicing contrast for stop consonants. The present experiment employed synthetic disyllables in an attempt to specify the contribution of the duration of syllable-initial and syllable-final formant transitions to the perception of stress. Results indicated that (1) such transitions do contribute to the durational percepts underlying stress judgments and (2) the extent of this contribution varies with the location of the transitions in the disyllable. [Work supported by NICHD.]

**NN6. Temporal and spectral cues to syllable stress.** Mary R. Smith (Department of Linguistics, UCLA, Los Angeles, CA 90024)

The interaction of temporal and spectral information for judgments of syllable stress was studied using a formant pattern of each of the vowels /i/, /I/, and /æ/ of English. The patterns were arranged to form all nine possible disyllables in the nonsense frame /s-ds-d/. In all disyllables the duration of the first vowel varied in 10-ms steps from 50 to 150 ms while the duration of the second vowel was changed to keep the work length constant. Fundamental frequency and intensity were kept constant within and across the words. Eight subjects judged the stress, length, and phonetic identity of the target words in isolation. The 50% crossover points for the stress and length judgments as a function of vowel duration were obtained by PROBIT analysis and subjected to an analysis of variance with phonetic identity as the treatment. Further tests show a significant difference between the locations of stress boundaries for different vowel pairs. The results are not so clearly a function of the inherent duration of the phonemes involved as previous research suggests.

**NN7. Perception of anticipatory coarticulation.** James G. Martin (Department of Psychology, University of Maryland, College Park, MD 20742)

Articulatory and acoustic studies of speech have shown that anticipatory coarticulation can reach across several segments in an utterance. However, the results of earlier perceptual studies suggest that perceptual effects are limited to adjacent segments only. In the present experiments, the stimuli (e.g., "I say tuzi") were of the form "I say C V<sub>1</sub> z V<sub>2</sub>" in which C was /p, t, or k/, V<sub>1</sub> was (u, æ, / and V<sub>2</sub> was /i, a/ in all 12 combinations, spoken by ten speakers. Reaction time (RT) of practiced listeners was observed to the final vowel target (V<sub>2</sub>). Sentence pairs differing only in final target vowel were either (a) computer cross-spliced at /z/ onset or (b) left intact (as spoken). In general, RT was slower to crossed targets, suggesting misleading information prior to the cross. These interference effects were dependent on prior phonetic context and differed between and within speakers. Some of the perceptual results were correlated with results from acoustic analyses of the sentence waveforms. It appears that anticipatory coarticulation across several segments provides information which can be used in perception. [Work supported by NINCDS.]

**NN8. Differences in production and perception of VOT in Polish and English.** Patricia Keating (36-521, Massachusetts Institute of Technology, Cambridge, MA 02139)

English and Polish use different Voice Onset Time (VOT) categories to contrast voiced and voiceless stops: short lag and long lag (English) versus prevoiced and short lag (Polish). When VOT is measured for word-initial stops in minimal pairs, sentences, and conversation, the VOT distributions for voiced and voiceless stops are clearly separated in Polish, but not always in English, especially in casual speech [Lisker and Abramson, *Language and Speech* 10, 1-28 (1967); Moslin, Brown University dissertation (1978)]. The per-

ception of VOT in synthetic stimuli was compared with production data for Polish and American English listeners. Both groups labeled stimuli in three continua that varied in range of VOT: -20 to +80, -100 to +50, and -100 to +20 ms VOT. The Polish, but not the American, listeners showed a mean boundary shift (from +20 to +4 ms VOT) as a function of these ranges. Thus the Polish VOT perceptual categories are somewhat unstable and do not always match the VOT production categories, although the latter are well-separated. In contrast, the English perceptual categories are quite stable, but the production categories sometimes overlap in VOT. Distinctiveness in production may compensate for instability in perception, and vice versa.

**NN9. Learning to recognize foreign speech sounds: Strategies of auditory processing.** David R. Lambert (2910 Edell Place, San Diego, CA 92117)

One can use two fundamentally different strategies to learn to recognize foreign speech sounds: Accept a sound as something new to be learned and form new codes which include accurate detail to represent it [Code Forming (CF) strategy], or assume the sound is a distorted version of something familiar, recognizing the sound immediately by using existing codes and ignoring inaccurate detail [Code Using (CU) strategy]. The language learning environment of the child or of a person in a foreign country encourages the CF strategy, while that of the typical classroom student encourages the CU strategy. Surprisingly, the two strategies result in about the same rate of learning. But they produce different processing capabilities. the CF strategy produces about 90 ms faster encoding; the CU strategy about 45 ms faster memorial comparison. In learning a language, one should use the CF strategy to develop fast encoding speed, while consciously focusing on other aspects of the language. (Reference: dissertation of same title, Pp. 330, 1977, PB-273 161/OGA.) [Work performed at Psychology Department, University of California at San Diego; supported in part by NIH.]

**NN10. Palato-alveolar affricates in several languages.** Ian Maddieson (Phonetics Laboratory, Department of Linguistics, UCLA, Los Angeles, CA 90024)

A project to examine phonetic differences between affricates is in progress. Voiceless palato-alveolar affricates and the most similar stops and fricatives have been recorded in closely matched medial positions from 10 speakers of each of three languages speaking at two different speech rates controlled by a metronome. The languages are (Mexican) Spanish, (British) English, and Italian. Spanish does not have a palato-alveolar fricative; English distinguishes affricate, stop, and fricative; Italian distinguishes single and geminate affricates and stops and has a palato-alveolar fricative. The general hypothesis under investigation is that such differences in the number of sounds which must be distinguished have implications for the ways in which the contrasts will be phonetically realized in the three languages, for example, in the variability found between speakers and between speech rates. Preliminary measurements on durational aspects of the segments examined indicate that there are differences between the languages which can be related to the general hypothesis. [Work supported by NSF.]

**NN11. Differentiation of plain and pharyngeal fricatives in Libyan Arabic.** Laurence J. Krieg, Peter J. Benson, and J. C. Catford (Department of Linguistics, The University of Michigan, Ann Arbor, MI 48109)

The considerable inventory of contrasting Arabic fricatives presents some hitherto unanswered questions to phonetic investigators: the acoustic cues differentiating the contrasts have only been partially specified [e.g., al-Ani, *Arabic Phonology* (The Hague, 1970)]; it has not been satisfactorily demonstrated whether fricative noise itself or only the surrounding vowels differentiate between corresponding pairs of pharyngealized ("emphatic") and nonpharyngeal-

ized fricatives [e.g., Jakobson, "“Mufaxxama” . . .,” *Selected Writings* 1, 510–522 (1962)], Bonnot [“Recherche experimentale sur la nature des consonnes emphatiques de l’arabe classique,” *Trav. Inst. Phonet. Strasbourg* 9, 47–88 (1977)]. A dialect of Libyan Arabic which contrasts an unusually large number of plain versus pharyngealized fricatives is recorded; measurement is made of time-domain variables such as VOT, rise time, and relative amplitude of the fricative. Spectral information is averaged together to provide representative spectra of each of the contrastive fricatives. These spectra are then transformed using various weighting functions; the original, the transformed spectra and time-domain measurements are then compared and grouped by multivariate analysis techniques to provide evidence for the general phonological grouping of fricatives in this dialect and the role of pharyngealized/plain contrast.

**NN12. On the recognition of the Spanish fricatives /s/ and /f/. Jorge Gurlé (Laboratory of Sensory Research, Buenos Aires, Argentina)**

The present work examines the perceptual load carried by the relative intensity of the noise to the intensity of the vowel in the identification of the Argentine Spanish fricatives /s/ and /f/. The stimuli consisted of synthetic fricative-vowel syllables. The fricative portion, consisting of a fixed band noise with a central frequency of 4500 Hz and a bandwidth of 300 Hz, was varied in its relative amplitude to the vowel, to obtain a set of ten noise amplitudes in relation to the fixed vowel amplitude. Spanish listeners identified low amplitude noises as /f/ and high ones as /s/. In a second experiment, subjects were tested on the same stimuli at two different overall sound-pressure levels of the syllable: 60 and 80 dB. For the 80-dB level, subjects identified more /f/ sounds than at the 60-dB level. This result can be interpreted as a different interaction between the vocalic portion and the fricative noise. It is suggested that this kind of interaction may be due to a differential masking effect of the vowel at different amplitudes.

**NN13. Noise intensity cues of spoken fricatives. George P. McCasland (University of Pennsylvania, 4045 Baltimore Avenue, C-3, Philadelphia, PA 19104)**

In spoken fricative-vowel syllables /fi, fa, fu, θi, θa, θu, si, sa, su, ji, ja, ju/, the fricative noise was replaced by segments of both /f/ and /θ/ sustained noise the intensity of which had been amplified to –6 dB relative to vowel level; 24 modified syllables in all. *Sibilant* /s/ percepts were heard in nearly 100% of the originally /s/ and /f/ syllables and in the majority of the originally /f/ and /θ/ syllables; *nonsibilant* /f/ and /θ/ percepts in the remainder. It was previously shown (Harris, 1958) that when /f/ and /θ/ noise at the natural, low intensity level relative to the vowel replaced the fricative noise in originally /s/ and /f/ syllables, /θ/ percepts resulted. In the present experiment, relatively high intensity /f/ and /θ/ noise served as a cue for /s/ percepts, typically overrode conflicting formant transition cues and might be considered *strident* in distinctive feature terminology. Noise spectrum is implicated only in the lack of *sibilant* /j/ percepts. This investigation extends work previously reported by the author [J. Acoust. Soc. Am. Suppl. 1 63, S21(A) (1978) and 65, S78(A) (1979)].

**NN14. Psychoacoustic parameters of the /r/-/l/ perceptual distinction. A. K. Syrdal-Lasky (Callier Center, University of Texas at Dallas, 1966 Inwood Road, Dallas, TX 75235)**

The possibility that the psychoacoustic distinction between rising and steady-state *F3* transitions may underlie the categorical perception of /r/ and /l/ was raised in a study of perception of the /ra/-/la/ continuum [Miyawaki, Strange, Verbrugge, Liberman, Jenkins, and Fujimura, *Percept. Psychophys.* 18, 331–340 (1975)], but later rejected because the perceptual distinction between /ri/ and /li/ was made between two stimuli with raising *F3* transitions rather than

between rising and steady-state *F3* stimuli [McGovern and Strange, *Percept. Psychophys.* 21, 162–170 (1977)]. The possibility that the perceived distinction between rising and steady-state *F3* occurred further from the physical steady-state value because of masking by the spectrally proximal *F2* in /ri/-/li/ syllables was examined and confirmed using five synthetic stimulus conditions: /ri/-/li/ syllables identical to McGovern and Strange’s, /ri/-/li/ syllables composed of *F2* and *F3* only, /ri/-/li/ syllables with *F3* amplitude raised and *F2* amplitude lowered, a /ri/-/li/ duplex condition, and isolated *F3* stimuli. The perceptual distinction between /ri/ and /li/ was made closer to the physical distinction between rising and steady-state *F3* as masking by *F2* was reduced. [Work supported by NSF.]

**NN15. Distinctive features of sibilants: A closer look. J. C. Catford and L. J. Krieg (Department of Linguistics, University of Michigan, Ann Arbor, MI 48109)**

Perkell, Boyce, and Stevens [*Speech Communication Papers Presented at the 97th Meeting of the Acoustical Society of America*, 109–113 (1979)] emphasized the quantal role of the sublingual cavity in differentiating the noise spectrum of the sibilants [s ʃ] in English. Other languages, however, differentiate more than two primary articulatory postures for sibilants. For example, there are three in such languages as Polish, Marathi, and Basque. The greatest number of primary contrastive postures for sibilants appears to be four, as in three Northwest Caucasian languages (Adyghe, Ubykh, and Bzyb). Previously published radiographic tracings for these languages, together with our spectral analyses suggest the need for either a multivalued sublingual cavity feature (nil, small, medium, or large, together with associated frequency characteristics) or additional articulatory distinctions (such as *alveolar* versus *postalveolar* and *apical* versus *laminal*).

**NN16. An INDSCAL analysis of normal and hearing-impaired listeners’ consonant confusions on a nonsense syllable test. J. L. Danhauer (Department of Speech, University of California-Santa Barbara, Santa Barbara, CA 93106), K. J. Doyle (University of Southern California, Los Angeles, California 90007), and B. J. Edgerton (Walt Disney Hearing Rehabilitation Research Center, Los Angeles, CA 90028)**

Recently, several authors have posited perceptual feature systems to describe consonantal relationships judged by both normal-hearing and hearing-impaired listeners. While these studies have enhanced knowledge of feature usage by such listeners through confusion, similarity, and dissimilarity paradigms, few clinical applications have resulted. Lately, there has been some revived interest in the use of nonsense syllable stimuli for clinical speech discrimination testing. The purpose of this study was to evaluate feature usage by both normal and hearing-impaired listeners to the pre-vocalic consonants of a nonsense syllable test (NST). Listeners were ten normal and eight hearing-impaired adults. The NST consisted of two 25-item lists of CVCVs randomly constructed from 22 consonants and 10 vowels of English. Listeners were tested individually under phones at ascending presentation levels of 5, 15, 25, 35, 45, and 55 dB (re: their SRTs). Responses were phonetically transcribed by the experimenter using both auditory and visual cues. Confusion matrices for both listener types and lists were grouped by presentation levels and submitted to INDSCAL for evaluation in five-dimensional solutions. Results of perceptual features recovered are presented and discussed in light of those found in earlier investigations.

**NN17. The contribution of synthetic low-frequency speech information to speechreaders. LeeAnn H. Ardell, Patricia Kuhl, and David W. Sparks (Speech and Hearing Science Department, University of Washington, Seattle, WA 98195)**

The contribution of frequency and amplitude information carried in the fundamental frequency (*f*<sub>0</sub>) of speech to the speechreading performance of normal-hearing subjects was measured using a con-

nected-discourse tracking procedure [C. L. DeFilippo and B. L. Scott, *J. Acoust. Soc. Am.* **63**, 1186–1192 (1978)]. Tracking performance was measured under five receiving conditions: (1) speechreading alone, (2) speechreading in combination with lowpass-filtered (below 300 Hz) speech (LPF), (3) speechreading in combination with the  $f_0$  extracted from the speaker's voice and presented at a constant amplitude (FM alone), (4) speechreading in combination with a 225-Hz pure tone (the mean  $f_0$  for this speaker), amplitude-modulated by the intensity variations occurring in the lowpass-filtered version of the speech waveform (AM alone), and, (5) speechreading in combination with the extracted and amplitude-modulated  $f_0$  (AM + FM). Performance was highest for LPF and AM + FM and lowest for speechreading alone with AM alone and FM alone performance in between. The implications of these results for the development of auditory and tactile prostheses for the profoundly hearing impaired will be discussed. [Work supported by NIH.]

**NN18. Tests for categorizing the auditory speech discrimination in the severely/profoundly deaf young adult.** J. C. Webster (Department of Communication Research, National Technical Institute for the Deaf, Rochester Institute of Technology, Rochester, NY 14623)

The phonetic constant, word familiarity and grade level comprehension of the 1922 monosyllabic words used in ten free responses and eight rhyme tests were analyzed. Printed versions of the predictable SPIN sentences were given for 35 NTID students to fill the missing last (key) word. Based on analysis of these words three new tests were conceived. (1) A six-choice vowel initial and final consonant and numeral test was found to categorize 35 NTID students more efficiently than the present spondaic word/CHABA (CID) sentences profiling tests. (2) The SPINIKER test, a competing message test using the concepts (and many sentences) of SPIN to be used for speech (lip) reading and/or auditory discrimination testing. (3) A common-word more or less unlearnable question or sentence test using multiple choices among interrogatives (where, when, . . .), verbal auxiliaries (can, will, . . .), personal pronouns and free responses among verbs or nouns to be used in hearing aid evaluations. Typical sentences are "What could she drink?" or "Where is my bag?"

**NN19. Evaluation of the sensitivity of the speech perception in noise test to the linguistic and acoustic cues utilized in speech discrimination.** J. H. Owen (Hearing and Speech Department, Kansas University Medical center, Rainbow Boulevard at 29th Street, Kansas City, KS 66103)

The Speech Perception in Noise (SPIN) test [D. Kalikow, K. Stevens, and L. Elliott, *J. Acoust. Soc. Am.* **61**, 1337–1351 (1977)] is a sentence test which measures the contribution of linguistic cues to the discrimination score. This measure is known as the difference score. This study was conducted to determine if language skills are related to this score. Ninety subjects, ranging in age from 8 to 11 yr and from 18 to 79 yr, were divided equally into three groups: normal hearing/normal language, impaired hearing/normal language, impaired hearing/impaired language. Subjects were administered the SPIN (at 6 SNRs), language, and IQ tests. Results indicate that the difference score is not significantly related to linguistic skills, age, or IQ, but is related to hearing loss and SNR. Interpretation of results indicates, however, that the difference score is influenced by test sentence audibility.

**NN20. Use of optical information in phonetic perception.** Quentin Summerfield (MRC Institute of Hearing Research, University of Nottingham, United Kingdom)

The accuracy with which normal listeners can transcribe sentences presented 12 dB below a background of continuous prose was com-

pared with accuracy in four audio-visually supplemented conditions. With monochrome displays of the talker showing (i) the face, (ii) the lips, and (iii) four points at the centres of the lips and the corners of the mouth, accuracy improved by 43%, 31%, and 8%, respectively. In contrast, no improvement was obtained with an optical specification of syllabification by a Lissajous circle whose diameter varied according to the amplitude envelope of the test sentences. The differences suggest that optical concomitants of articulation specify linguistic information and do not merely focus attention when there is a competing background. This conclusion was reinforced in a second experiment, in which identification functions were obtained for continua of synthetic utterances. These ranged between /aba/, /ada/, and /aga/, and were presented both in isolation and in combination with videorecordings of a speaker uttering each of the disyllables corresponding to the endpoints. Percepts in audiovisual conditions differed from those in audio-alone conditions: audio-visually, /b/ was only perceived when lip-closure was specified optically, and, if lip closure was specified optically, /b/ was generally perceived. Interpretation of the detailed results suggests that perceivers make use of articulatory constraints on the combined audio-visual specification of phonetic events.

**NN21. Identification and discrimination of VOT by listeners with moderate, severe and profound sensorineural hearing loss.** Susan Parady, M. F. Dorman, P. Whaley (Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85281), and L. J. Raphael (Lehman College, CUNY, Bronx, NY 10468 and Haskins Laboratories, 270 Crown Street, New Haven, CT 06510)

To determine whether sensorineural hearing loss affects listeners' ability to detect the voice onset time (VOT) of stop consonants, listeners with moderate, severe, and profound hearing loss were presented stimuli from along a VOT continuum in both identification and discrimination tasks. The location of the phoneme boundary did not differ among the normal listeners, the listeners with moderate loss, and most listeners with severe loss. Some listeners with severe loss, however, evidenced longer boundaries while others could not identify the signals at all. A similar outcome was obtained for the listeners with profound loss. For the most part discrimination data mirrored identification data. However, in several instances listeners who were unable to identify voiced and voiceless stops were able to discriminate between them in a normal manner. Since identification and discrimination tasks assess different aspects of information processing, an evaluation of sensory capabilities in the hearing impaired should include both tasks.

**NN22. Intelligibility of spoken letters of the alphabet.** A. E. Rosenberg, C. L. Geter, B. J. McDermott, K. L. Shipley, and J. G. Wilpon (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

A subjective experiment was carried out to assess the effect of spelling context on the ability of listeners to identify spoken letters of the alphabet. Individual letters, spoken by a male talker and a female talker, were concatenated to form strings of up to ten letters representing the spelling of a last name (surname) and one or two initials. Three spelling contexts were observed. First, real names were selected at random from the 20 000-entry Bell Labs telephone directory. Second, strings were constructed using trigram statistics for letter sequences tabulated from the same directory. Third, letters were selected at random to form strings. In addition, three masking noise conditions were observed: no noise, low-level, and high-level broadband additive masking noise. The results show that the listeners made fewer errors identifying letters imbedded in real name or trigram strings than in random letter strings. The effect is especially pronounced in the presence of masking noise. Confusions among the spelled letters are also tabulated.

**Session OO. Additional Papers (Poster Session)**

Abstracts of contributed papers will be available in the registration area.

**OO1. Examples of optimized arrays.** Barry G. Buehler and Roy L. Streit (Naval Underwater Systems Center, New London, CT 06320)

**OO2. Comparison of SEA and Handbook Analysis Procedures for a Noise Propagation problem.** C. I. Holmer and B. L. Burkewitz (Noise Control Technology, Division of Underwater Systems, Inc., Silver Spring, MD 20910)

**OO3. Acoustic markers of personality and emotion.** Bruce L. Brown (Department of Psychology, Brigham Young University, Provo, UT 84602)

**OO4. A weighted overlap-add method of short-time Fourier analysis/synthesis.** R. E. Crochiere (Acoustics Research Department, Bell Laboratories, Murray Hill, NJ 07974)

**OO5. Identification of nasal and liquid consonants spliced from Spanish syllables.** M. I. Massone (Laboratorio de Investigaciones sensoriales, CONICET, CC53, Suc. 53, 1453 Buenos Aires, Argentina)

**OO6. Selective measurement of impaired frequency resolution by a 4-alternative speech feature test (FAAF).** M. P. Haggard, J. R. Foster, and R. S. Tyler (MRC Institute of Hearing Research, Nottingham NG7 2UH, England)