Program of the Sixty-First Meeting of the Acoustical Society of America

BELLEVUE STRATFORD HOTEL, PHILADELPHIA, PENNSYLVANIA, MAY 10–13, 1961

Session A. Physiological Acoustics

ROBERT C. BILGER, *Chairman*

Contributed Papers

A1. Cochlear Potentials in the Human Being. ROBERT J. RUBEN, JOHN E. BORDLEY (nonmember), AND ALFRED T. LIEBERMAN, *Department of Otolaryngology, Johns Hopkins University, Baltimore 5, Maryland.*—For many years there have been various attempts to record cochlear potentials from the round window in a human being. Some earlier efforts have met with moderate success. During the past two years, approximately 30 cases have been attempted at the Johns Hopkins Hospital. A multichannel tape recorder has been used to simultaneously stimulate the human ear and record from the round window. The patients all had their tympanic membranes reflected and a small electrode was placed on the round window niche. In most cases, good cochlear potentials were recorded and measured. In addition, clicks were used to stimulate the human ear, and recordings have been made from the round window. The action potentials of the 8th nerve known as $N1$ and $N2$ were then recorded. Two groups of patients are now being studied. The first group consists of patients with otosclerosis. A recording is made before mobilization or a replacement of the stapes is done. A second recording is taken after the oval window is free, to show if there has been any objective gain in the cochlear potential. The second group of patients are children with congenital neurosensory hearing loss. Excellent cochlear potentials and 8th nerve action potentials have been recorded in this group. The recording of cochlear potential in man is being used as a diagnostic and exploratory tool in evaluation and understanding of hearing disorders. It is felt that this also gives a valuable tool in understanding some of the basic properties of hearing in the human being.

A2. Properties of the 8th Nerve. ROBERT J. RUBEN, HUGO FISH (nonmember), AND WILLIAM HUDSON, *Department of Otolaryngology, Johns Hopkins University, Baltimore 5, Maryland.*—There has been some debate as to whether or not the action potential of the 8th nerve as viewed at the round window, and $N1$ and $N2$, originate within the cochlear or in the nervous system. Several experiments have been undertaken in cat in which the 8th nerve was sectioned completely, leaving the blood supply intact. By recording from the severed distal stump and the round window, the $N1$ and $N2$ were found to be preserved up to 24 hr after sectioning. By using careful time studies, several properties of the 8th nerve were also noted. The latency of the $N1$ as recorded at the round window with the first potential recorded on the 8th nerve was carefully measured. A discrepancy in conduction time was found. The conduction time was based upon the fiber diameter which has been measured, and also direct studies. The knowledge gained from these experiments will be of great use in the interpreting of the cochlear response to click as recorded in the human round window. The locus of both the $N1$ and $N2$ has been shown to be peripheral to the cochlear nucleus. Time studies indicate that the origin of the $N1$ and $N2$ is at some point peripheral to the ganglion cells in the modiolus.

A3. Auditory Nerve Responses to Low-Pass Transients. D. C. TEAS, D. H. ELDREDGE, AND H. DAVIS, *Central Institute for the Deaf, St. Louis 10, Missouri.*—Masking has been used to subtract groups of nerve impulses from the whole-nerve action potential of the guinea pig. The form of that part of the whole-nerve action potential which is eliminated by a narrow band of noise is approximately diphasic. This is the expected form of individual nerve impulses recorded from one electrode in the cochlea and the other on the neck. The stimulus variables investigated were intensity and rise time of the acoustic transient, and intensity, bandwidth, and center frequency of the masking noise. The whole-nerve action potential to low-pass transients is composed of impulses that arise from a large extent of the organ of Corti, with a distribution in time that parallels the space-time pattern of the cochlear microphonic. (This research was supported by Bt-366-C2 and NIH-B1726.)

A4. Acoustic Trauma: Concomitant Behavioral and Electrophysiological Measurements in the Cat. DANA BEATTY, *Division of Speech Pathology and Audiology,* AND F. BLAIR SIMMONS (nonmember), *Division of Otolaryngology, Stanford University, School of Medicine, Palo Alto, California.*—Cats with electrodes permanently implanted on the round window of the cochlea and previously trained to respond to threshold and supra-threshold tones in shuttle-box conditioning were exposed to traumatic sound (135 db at 1 kc for 2 hr). Serial concomitant measurements of both behavioral and electrophysiologic (CM and AP) responses were obtained before and after exposure at several frequencies. Temporary threshold shifts of 65 to 95 db were produced in the middle frequency range. Associated cochlear microphonic changes were considerably less (5 to 25 db) and in most cases returned to pre-exposure values long before behavioral indexes at varying degrees of permanent threshold shift.

A5. Minimum Phase Responses for the Basilar Membrane. JAMES L. FLANAGAN AND CAROL M. BIRD (nonmember), *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Appreciable experimental data exist for the amplitude of basilar membrane displacement as a function of sound frequency. Data on the phase of displacement vs frequency are relatively meager. At low frequencies the latter do not relate well to the phase predicted by recent mathematical models for membrane displacement [Bell System Tech. J. 39, 1163–1191 (1960)]. To examine the difference, we programmed an IBM-7090 computer to calculate: (a) minimum phase functions corresponding to the experimental amplitude vs frequency responses; (b) inverse Fourier transforms of the experimental amplitude and phase data; and (c) inverse transforms of the experimental amplitude and calculated minimum phase data. The results were: (a) At low frequencies the calculated phase differs from the experimental phase by about $\pi/2$ rad or more, with the membrane displacement leading that of the stapes. Constant delay (i.e., linear phase) does not account for the difference; (b) inverse transforms of the experimental data yield nonrealizable impulse responses, i.e., responses which are nonzero for negative times; and (c) inverse transforms of the experimental amplitude and calculated phase yield proper impulse responses. At low frequencies, therefore, the experimental amplitude and phase data do not seem compatible. Because of the great difficulties attending the experimental measurement, it is the phase data which are thought to need confirmation.

A6. An Equivalent Circuit Study of Fenestration. G. F. PASKUSZ, *Department of Engineering, University of California at Los Angeles, Los Angeles 24, California.*—This paper describes an attempt to supplement the experimental approach to improving fenestration by circuit analysis. The fenestrated ear is considered an idealized acoustical system and its equivalent circuit is derived. The circuit consists of an input section (outer ear, middle ear, fenestrae), and seven sections representing the inner ear. These are found to be resonant at frequencies between 100 cps and 10 kc. The problem is simplified by assuming the partition between outer and middle ear to be rigid. The other significant parameters are then combined into a variable series capacitance. The resulting circuit is analyzed numerically (digital computer). Response curves are obtained for volume displacement across the basilar membrane as a function of frequency and distance from the input end, with input capacitance as parameter. The computed results and experimental findings (by Békésy) are similar. The startling result of the investigation is that a lowering of the combined window stiffness by a factor of 30 results, in the model, in a 30 db gain over most of the speech spectrum. This is in sharp contrast to controversial experiments in the 1930's by Hughson and Crowe who tried to improve fenestration by increasing the stiffness of the round window.

Session B. Underwater Acoustics—Propagation Phenomena

J. C. JOHNSON, *Chairman*

Invited Paper

B1. Reverberation in the Ocean. HAROLD L. SAXTON, *U. S. Naval Research Laboratory, Washington, D. C.*—The problems associated with acoustic reverberation in the ocean continue to be of prime importance in the use of active sonar. Even though a great amount of effort has been devoted to these problems, they are still not well understood and experimental evidence remains the prime source for an evaluation of the state-of-knowledge, in this field. This paper attempts to summarize the state-of-knowledge through the presentation of recognized fundamentals and results of pertinent experiments.

Contributed Papers

B2. Correlations and Frequency Spectra of Fluctuations in Sound Signals Transmitted over a Fixed Path in an Estuary. J. R. SMITHSON, *Chesapeake Bay Institute, The Johns Hopkins University, Baltimore 18, Maryland.*—Short pulses of 100-kc sound have been transmitted from a fixed source to an array of fixed receivers at a range of 100 ft in the Severn River. A statistical analysis has been made of 7000 pulses transmitted directly across a sharp, changing velocity gradient, which produced large fluctuations, and of 14 000 pulses reflected from the surface when fluctuations in direct transmission were negligible. Autocorrelations and cross correlations with delays up to 300 sec have been computed for signals received simultaneously on hydrophones with various horizontal and vertical separations. For surface-reflected signals, little correlation was found between successive pulses or between pulses received on separated hydrophones. For direct transmission, the decrease of autocorrelation with time and the decrease of cross correlation with both time and distance have been compared with various theoretical correlation functions. Phase differences between fluctuations on separated hydrophones have been found which are not simply related to the current. Frequency spectra of the fluctuations have been computed. In surface-reflected signals there is fluctuation at frequencies up to 6 cps, while in direct transmission there is negligible fluctuation at frequencies above 0.2 cps. (This work was supported by the Office of Naval Research.)

B3. Observations of the Stability of a Normal Mode Sound Field in an Intermediate Scale Model. J. A. SCRIMGER (nonmember), *Pacific Naval Laboratory, Esquimalt, British Columbia.*—Spectra of sound pressure amplitude versus frequency in the range 1.1–2.4 kc have been obtained at regular intervals in shallow (10 ft) water during two periods of extended observation. The acoustical data have been presented in the form of contour diagrams of sound pressure amplitude versus frequency and time as coordinates and compared with water conditions. The form of the contour diagrams permits differentiation between field structure variation and variation in attenuation, although differentiation between the associated processes has not been possible. Temporal variation in the vertical sound velocity profile produced only small changes in the gross features of the field structure—as inferred from the contour diagram—for up to the first three modes of propagation. For frequencies above 1.6 kc (i.e., when three or four modes were stimulated) the variations in field structure were prominent. Attempts have been made, with little success, to explain periods when the received signal was strongly attenuated in terms of vertical sound velocity structure.

B4. A Mathematical Definition of Acoustic Coherence. A. SHAPIRO (nonmember), *Bell Telephone Laboratories, Whippany, New Jersey.*—The concept of coherence in the field of acoustics apparently means the same thing to all people, but only on an intuitive level, and a formal definition has not been generally agreed upon. The author suggests that the definition used in the theory of multidimensional stochastic processes be adopted by the acousticians. In the present paper, this definition is presented and, when Gaussianity is assumed, an interpretation is given in terms of the relationship between the phases of the signals. The proposed definition leads to the following interpretations: when the processes are highly coherent, the variability of the phase difference between the signals is small; when the processes are incoherent, the phase difference has a high degree of variability. This concept of coherence seems to agree with the intuitive notions currently used in acoustics. (This work was supported by the Office of Naval Research.)

B5. Measurement of Acoustic Coherence Coefficients in Deep Water. G. H. Robertson (nonmember), *Bell Telephone Laboratories, Inc., Whippany, New Jersey.*—Coherence measurements have been made relating sinusoidal signals simultaneously received at separated hydrophones. The definition of coherence advocated by Shapiro in the previous paper is used. The quantity measured is closely related to phase. Of interest is long term average phase as well as a measure of the fluctuations about this mean. Such measurements are useful as an aid in understanding the performance of complex hydrophone systems. The measurement is accomplished by cross multiplying and averaging stored five-second segments of information from each of two channels. This computation is carried out for a range of delays comprising several periods of the sinusoidal signal. A facsimile type recorder provides a pictorial display of how phase wanders as a function of time. Long term averages are obtained numerically from samples of the continuous display. Recently collected quantitative data are presented for several hydrophone spacings in deep water. The results show violent phase fluctuations and dramatically illustrate the statistical nature of acoustic transmission in the ocean. (This work was supported by the Bureau of Ships.)

B6. Acoustic Propagation in a Two-Layered Model—Transverse Waves in Bottom. R. K. Eby, *Polychemicals Dept., E. I. duPont de Nemours and Co., Inc., Wilmington, Delaware,* and A. O. Williams, Jr., *Department of Physics, Brown University, Providence 12, Rhode Island.*—The complex shear modulus has been measured for samples of Hycar rubber under these conditions: (a) 0.25–2.0 cps at 22°C, with a torsion pendulum; (b) 40 cps–5 kc, between 0° and 41°C, with an electromagnetic transducer; (c) at 10 Mc and 22°C, by McSkimin's method. The "method of reduced variables" allows calculation of the shear modulus, from (b), from 5 cps to 1.2 Mc at 22°C. Measurements (a) and (c) are consistent. From the density and the real part of the modulus, the speed of transverse waves in the rubber has been calculated as a function of acoustic frequency. Results are compared with speeds inferred in a previous paper [R. K. Eby, A. O. Williams, R. P. Ryan, and P. Tamarkin, J. Acoust. Soc. Am. 32, 88 (1960)] that treated acoustic propagation in a model made up of a shallow water layer over a thick slab of this same rubber. The present speeds fall about 25% below those previously inferred, are undoubtedly more reliable, and are still reasonably successful in explaining attenuations within the water layer caused by transverse waves in the bottom, as hypothesized earlier. (A. O. W., Jr., assisted by the Office of Naval Research.)

B7. Acoustic Attenuation in a Liquid Layer over a "Slow" Solid. A. O. Williams, Jr., *Department of Physics, Brown University, Providence 12, Rhode Island.*—Ewing and Press have discussed acoustic propagation in a liquid layer, over a solid with both compressional wave speed c_2 and transverse wave speed c_t exceeding c_1 for the liquid. When, instead, $c_2>c_1>c_t$, sound is only partially trapped in the liquid layer, with steady "leakage" into transverse waves. Such a case has been measured in a laboratory model (water over rubber) [R. K. Eby, A. O. Williams, R. P. Ryan, and P. Tamarkin, J. Acoust. Soc. Am. 32, 88 (1960)]; perhaps a compact sand bottom acts similarly. The problem is solved by satisfying acoustic boundary conditions at the interface, for a transverse and two compressional waves; when c_t vanishes the equations reduce to the liquid-liquid (Pekeris) case. The eigenvalue for each normal mode is now complex, betokening "leakage." The real part is altered (relative to liquid-liquid) as if the density of the bottom were decreased. The imaginary part gives an attenuation coefficient, for sound traveling into the liquid layer, roughly proportional to $(c_t/c_1)^3$, with a complicated dependence on other parameters. The calculation can be extended to include absorption of transverse waves in the bottom—probably the usual situation in a "flabby" material. (Work supported in part by the Office of Naval Research.)

Session C. Waves and Vibration

Joshua Greenspon, *Chairman*

Contributed Papers

C1. Determination of Finite Amplitude Distortion by Light Diffraction. B. D. Cook, *Department of Physics, Michigan State University, East Lansing, Michigan.*—The theory of a new method of determining the harmonic structure of a finite amplitude wave by light diffraction has recently been given [B. D. Cook, J. Acoust. Soc. Am. 32, 336 (1960)]. This method allows the computation of the amplitudes of the harmonic components of the instantaneous pressure from the measurements of the light intensities of all of the diffraction orders. Preliminary measurements and computations indicate that this method will be a valuable tool in the study of finite amplitude distortion. (This work was supported by the Office of Ordnance Research, U. S. Army.)

C2. Diffraction of Wide and Narrow Light Beams by Distorted Finite Amplitude Progressive Ultrasonic Waves. L. E. Hargrove, *Department of Physics, Michigan State University, East Lansing, Michigan.*—Results of an investigation of the diffraction of light passing through an ultrasonic wave of finite amplitude are given. A method is developed and used to determine the second harmonic component of the distorted ultrasonic wave at various distances from the transducer, while maintaining a constant local fundamental component. A wide light beam (giving discrete diffraction orders) is used for these determinations. It is then shown that the observed continuous light distribution (using a narrow light beam) is in good agreement with the theoretically predicted distribution. The effects of third and higher harmonics are neglected. These determinations of second harmonic are compared with values obtained from diffraction effects observed after passing the second harmonic component through an acoustic filter plate. Measurements were made at 3.0 Mc in water with approximately $\frac{3}{4}$ of an atmosphere local ultrasonic pressure amplitude. (This work was supported by the Office of Ordnance Research, U. S. Army.)

C3. The Scattering of Sound by a Thin Prolate Spheroidal Shell. Alexander Silbiger, *Cambridge Acoustical Associates, Inc., Cambridge 38, Massachusetts.*—This paper deals with the scattering of a plane wave incident on a prolate spheroidal shell along its axis of revolution. The procedure can be easily generalized to an arbitrary direction of incidence. Contrary to the case of spherical and cylindrical shells, the spheroidal wave functions in terms of which the incident and scattered waves are expressed, do not coincide with the normal modes of the shell. As a result, when the coefficients of the expression of the shell motion in the spheroidal functions are used as generalized coordinates, all the Lagrangian equations will be coupled. Expressions suitable for computer calculations are given, and an example is worked out in which the frequency of the incident wave is close to one of the natural frequencies of the shell. Under these conditions the scattering action of the shell differs strikingly from that of a rigid spheroid. (This paper is based on work sponsored by the Office of Naval Research.)

C4. Driving Point Impedances of Cylindrical Shells. W. THOMPSON, JR., AND J. V. RATTAYYA (nonmember), *Cambridge Acoustical Associates, Inc., Cambridge 38, Massachusetts.*—Explicit expressions for the driving point impedance of cylindrical shells subjected to concentrated fluctuating radial forces have been sorely lacking in the extensive literature on shell vibrations. This need was partly filled by a recent paper by P. A. Franken [J. Acoust. Soc. Am. 32, 473 (1960)], but his impedance expressions have been found to be valid only over a restricted frequency range. The purpose of the present paper is to supply convenient, more generally applicable expressions for cases of practical interest. The loads considered are sinusoidally varying, concentrated and locally distributed forces and moments. Impedance expressions are given for uniform shells, both infinite and finite. Impedance expressions are also given for the ring-stiffened shell of finite length. (This paper is based on work sponsored by the Office of Naval Research.)

C5. Measurement of Driving Point Impedance of Thin Plates. D. A. THOMAS AND T. A. HENRIQUEZ, *U. S. Navy Underwater Sound Reference Laboratory, Orlando, Florida.*—The driving point impedance of thin plates as implied by the work of Zener and given explicitly by D. A. Thomas [J. Acoust. Soc. Am. 30, 220 (1958)] was determined by measuring with a pulse method the complex force and velocity at the driving point. The plate was terminated in wet sand to simulate infinite plate conditions. At low frequencies, the measurements were limited by the physical dimensions of the plate. At high frequencies, compliance in the mounting of the accelerometer and driving coil caused inaccuracies in the measurements. Measurements were made on one aluminum plate and one steel plate. The data obtained agreed well with theory in the frequency range 300 cps to 3 kc.

C6. Some Measurements of the Absorption Coefficient of Soil Using the Impedance Tube Technique. JAMES H. PROUT, *Institute of Science and Technology, The University of Michigan, Ann Arbor, Michigan.*—In any field testing involving horizontal propagation of sound, it is desirable to know how much is absorbed by the ground. Since the sound absorption properties of the ground are affected by many factors, it is usually necessary to make an estimate of this effect. This paper describes an experiment to measure absorption coefficient of the soil by means of an impedance tube. Controlled laboratory tests are described which measured the effect of moisture content and particle size on the absorption coefficient. Effects of grass on absorption coefficient are also noted. (This work was sponsored by Project Michigan under a U. S. Army contract.)

Session D. Architectural Acoustics—Transmission Loss

JACK B. C. PURCELL, *Chairman*

Invited Papers

D1. Factors which Determine the Sound Transmission Loss of Walls. B. G. WATTERS, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Walls are commonly delineated by a cross-sectional drawing. Although a "one-dimensional" description may be sufficient for building the wall, it does not generally convey enough information from which to calculate accurately the transmission loss. For single walls, the size and the detail at the edges are important [M. Heckl and K. Seifert, "Untersuchungen über den Einfluss der Eigenresonanzen der Messräume auf die Ergebnisse von Schalldämmessungen," Acustica 8 (1958); and I. Dyer, WADD Tech. Rept. (to be published)]. These two factors will be discussed in detail in the following paper. For walls separating reverberant rooms, the relative volumes, the sound absorption, and the symmetry of the rooms are important [M. Heckl, "Die Schalldämmung von homogenen Einfachwänden endlicher Fläche," Acustica 10 (1960)]. For any wall, the angle of incidence of the sound wave is important [A. Eisenber, "Über die Schalldämmung von Glasscheiben und Fenstern," pp. 34, 35 in *Bauforschung im Wohnungsbau*, Schwenk and Company, Frankfurt-am-Main, Germany]. The transmission loss of a wall is seen to be a function of many parameters, only one of which is the composition of the wall cross-section. The role of these other parameters and their importance for the testing and use of walls will be discussed.

D2. Influence of Area and Mounting Conditions on the Transmission Loss of Single Walls. MANFRED HECKL, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Finite walls with small damping are subject to resonances that lead to amplitudes which are often much greater than the amplitudes calculated for infinite walls. These resonances are due to excitation of free waves at the wall boundaries. The free waves do not usually affect the TL of a limp wall because in this case the free waves are radiated only to a small extent. For a stiff wall, however, the free waves are very important. It will be shown that they are responsible for some effects that have been measured (see previous paper) but cannot be explained by theories that assume walls of infinite size. In particular, the influence of the area and of the bending wave transmission into adjacent structures will be discussed.

Contributed Papers

D3. Some Field Measurements of Building Constructions Having High Sound Transmission Loss. RONALD L. MCKAY (nonmember), *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Field measurements of the sound transmission loss provided by various complex building constructions are presented. The constructions are of the type frequently used in music schools and laboratory test facilities. Data are given on double masonry walls, on masonry walls with resiliently furred plaster, and on floating concrete floors. Some data are compared with similar measurements reported in the literature. Certain conclusions are drawn concerning practical design procedures.

D4. The Reliability of Field Measurements of Sound Isolating Partitions. PARKER W. HIRTLE (nonmember), *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Transmission loss measurements have been made on field installations of several partitions. Results of field measurements of the same components in several different installations are compared. Some comparisons are made with laboratory measurements of installations which attempt to duplicate field conditions and those which do not.

D5. Sound Isolation of Conventional Doors in Field Installations. WILLIAM R. FARRELL, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The presence of doors often seriously limits the achievable room-to-room sound isolation even when they do not occur in the common dividing partition. Room-to-room noise reduction measurements have been made of sound transmission paths through doors, including doors in the common dividing partition, doors separated by various lengths of corridor, and doors opening into larger spaces (secretarial pools, etc.). The noise reduction varies with the door construction, tightness of fit, spacing of the doors, size and absorption of the receiving room, and/or the intervening space. Measured data are presented and some preliminary conclusions drawn.

D6. Measurement of Sound Transmission Through Door Seals. HALE J. SABINE, *Physics Research Division, Armour Research Foundation, Chicago 16, Illinois.*—A small-scale test facility has been constructed for measuring the transmission of sound through slits typical of the crack around a door. The facility consists of an absorptive lined box of concrete block containing a sound source beamed toward the top. The box is covered with a pair of heavy sliding panels having a joint at the center line of the box. Samples of slit configurations, gaskets, etc., are inserted between the panels for measurement. The box is built in a reverberation chamber, and the sound power output W of the slit is measured together with the sound pressure p_0 in the box immediately adjacent to the slit. The effective radiating length L of the slit is about 2 ft. The measurements yield a quantity tentatively called the "slit constant" $K=W/L\ p_1^2$ where $p_1^2=p_0^2/2$ is the equivalent room average sound pressure for the source side. The quantity K can then be used to compute the equivalent transmission loss TL' of a slit or seal forming the perimeter of a panel of given dimensions. For a 3×7-ft door, TL', defined as the TL of a door which transmits the same total sound power as the perimeter seal, is about 8 db higher than 10 log $1/K$, if K is evaluated in feet. Typical data on slit and gasket configurations are presented.

D7. A Double Wall Noise Control Enclosure for an Impulsive Sound Source. ROBERT M. HOOVER AND LLOYD J. WILLIAMS (nonmember), *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The design of a group of double wall noise control enclosures on the upper floor of an existing building is described and some noise reduction measurements are reported. A design objective for these enclosures was to achieve more effective noise reduction than that provided by existing heavy, on grade, double wall, reverberant enclosures, but with reduced weight. This objective was achieved by a combination of a lightweight inner skin, a moderately large air space, a masonry exterior, isolation of the inner room floor slab, and the use of sound absorbing materials. Other noise control measures included sound retarding doors and windows, mufflers in the air supply and return passages, and flexible service connections to the rooms. Peak and octave band noise levels taken inside and outside the old and new enclosures are reported.

Session E. Speech Communications I

A. S. HOUSE, *Chairman*

Invited Paper

E1. Iso-Preference Method for Evaluating Speech Transmission Circuits. W. A. MUNSON AND J. E. KARLIN, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Subjective listening experiments have been conducted to establish a method for determining (1) contours of equal listener preference for a variety of speech transmission conditions in the audio space, and (2) a scale of magnitude of preference between contours based on preference difference thresholds in which the numbers are called TPU's (Transmission Preference Units). The scale is shown to satisfy the transitivity requirement even for widely different transmission conditions. Hence any two conditions with the same TPU number can be predicted to be equal in preference. The method involves only simple AB preference judgments for two brief samples of speech heard consecutively.

Contributed Papers

E2. Recent Progress in Formant Synthesis of Connected Speech. C. G. M. FANT, J. MARTONY (nonmember), U. RENGMAN (nonmember), AND A. RISBERG (nonmember), *Speech Transmission Laboratory, Royal Institute of Technology, Stockholm, Sweden,* AND J. N. HOLMES (nonmember), *Joint Speech Research Unit, British Post Office, Ruislip/Middlesex, England.*—The formant coded series synthesis system OVE II and associated function generator for converting manually traced parameter lines to time-variable control voltages have been improved in several respects. An associated project on the pole-zero spectra matching of speech sounds has provided optimal coding of available circuitry. Audible quality evaluations of the effects of various types of approximations have been undertaken to some extent.

E3. An Artificial Talker Driven from a Phonetic Input. JOHN L. KELLY, JR. (nonmember), AND LOUIS J. GERSTMAN, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—This paper describes a method of producing artificial speech from a phonetic input, i.e., symbols representing the names of phonemes corresponding to a given text are fed into a machine and the acoustic waveforms of connected speech emerge. The experimental work was accomplished on an electronic computer (IBM 7090), but the scheme is simple enough to permit realization with analog hardware. The talking machine program is divided into two parts. The first part simulates a more or less conventional resonance synthesizer of the tandem variety, requiring nine control signals; buzz intensity, hiss intensity, pitch, plus the center frequencies and bandwidths of three formants. Initially, this part of the program was used alone in experiments for which the inputs were detailed specifications of the control signals derived from spectrograms and physiological data, sampled at approximately three times the phonemic rate. Results from this phase were later combined with known results in speech perception to produce the rules used by the second program. That program accepts for its input the names of phonemes punched on IBM cards and produces the control signals to drive the resonant synthesizer.

E4. The Digital Vocoder as a Research Tool. WILLIAM H. MORRIS AND SANFORD E. GERBER, *Hughes Aircraft Company, Los Angeles 45, California.*—A modular block diagram of the Hughes 12-channel vocoder is shown and a functional description is presented. The advantages of a commercially built digital vocoder as a research tool, such as reliability, modular construction, and digital processing of the output signals, are discussed. A brief discussion of the sampling theorem in terms of spectral analysis is followed by a description of the techniques used in examining the dynamics of clear speech through use of the digital feature of the Hughes Vocoder.

E5. Intelligibility Study of the Hughes Digital Vocoder. SANFORD E. GERBER AND WILLIAM H. MORRIS, *Hughes Aircraft Company, Los Angeles 45, California.*—A discussion is presented of the method and procedure of intelligibility testing employing artificial speech. Several experiments are described in which the vocoder was tested under various conditions. Since the Hughes vocoder is a digital system, the bit rate employed is of considerable interest in terms of its effect upon received intelligibility. Word intelligibility studies were performed using three different bit rates, and the results of these studies are discussed. Moreover, studies were made of various conditions of filtering within the vocoder, and these results are presented. Finally, future studies and applications of the digital vocoder are forecast.

Session F. Underwater Acoustics—Transducers, Calibration, and Noise

H. E. NASH, *Chairman*

Invited Paper

F1. On the Design of Underwater Transducer Arrays. SAM HANISH (nonmember), *U. S. Naval Research Laboratory, Washington 25, D. C.*—This paper presents a summary of the state-of-knowledge and some of the unsolved problems facing today's transducer array designers. Of particular importance are the problems associated with the mutual interaction between elements, for element radiating face dimensions smaller than one-half wavelength, when the elements are hard-driven. This aspect of design problems will therefore be the major focal point of the presentation.

Contributed Papers

F2. Magnetostrictive Flexural Bar Transducers. RALPH S. WOOLLETT, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—Although the use of magnetostriction to excite flexural vibrations has had scarcely any practical applications to date, very promising transducers can be made by exploitation of this principle. A basic resonator of this type consists of two flexural bars joined together by common end supports. The bars are necked down where they join the end supports to provide a flexible section, by means of which supported-end boundary conditions are approximated. The resonant frequency of this structure is very close to that of the ideal end-supported bar. To provide for magnetostrictive excitation of flexure, each bar is slotted lengthwise, and windings are placed on each half of the bar formed by the slot. Subsequently, the slot is filled with a hard plastic. An equation for the electromechanical coupling coefficient of this structure has been derived, which includes the effect of slot width. If the ideal end-support condition is realized, the coupling coefficient will be 80% of the coupling coefficient of the magnetostrictive material. A coupling coefficient of 0.25 has been achieved with laminations of nickel-cobalt alloy. Calculations indicate that a surface intensity of 1 w/cm^2 in an underwater transducer should be achievable without exceeding the fatigue limit.

F3. Measurements on Lithium Sulfate Crystals and Lead Metaniobate Ceramic at 10 000 psi. CLAUDE C. SIMS, *U. S. Navy Underwater Sound Reference Laboratory, Orlando, Florida.*—Measurement of the electroacoustic characteristics of lead metaniobate and lithium sulfate as volume expanders in the frequency range 100 cps to 3 kc is described. Two identical stacks of crystals are placed in a small chamber at the same time. The measurement technique is based on the assumption that the change is acoustic impedance of the pressure chamber with with hydrostatic pressure is known, and that consequently any changes in the characteristics of two like crystals, one receiving, the other transmitting, can be determined by measuring the change in the output as a function of pressure. Also, if the electrical impedance of the crystals do not change, then, by reciprocity, any change in the sensitivity of the crystals would multiply the change in the receiver by the change in the transmitter.

The possibility of compensating errors is therefore eliminated. Changes in response characteristics measured under pressure by this method are less than ±0.5 db.

F4. Hydrophone Calibrator. CLAUDE C. SIMS, *U. S. Navy Underwater Sound Reference Laboratory, Orlando, Florida.*—A system to provide rapid, reliable, economical calibration of small hydrophones is described. The system is designed for production testing and field use in the frequency range of 100 cps to 3 kc. The hydrophone is placed in an open-ended column of water which is excited by an electrodynamic driver. Acoustic pressure in the tube is calculated from system parameters and compared to measured results for various types of hydrophones. Results agree with open-water data within ±1.0 db.

F5. Near-Field Investigation of a Shaded Transducer. D. L. BAIRD AND C. M. MCKINNEY, *Defense Research Laboratory, The University of Texas, Austin 12, Texas.*—An investigation has been made of the near-field phase and pressure amplitude distribution across the aperture of a shaded line-in-cone transducer. The transducer consists of a line of 16 cylindrical elements mounted within and along the axis of a right circular conical reflector. Data were taken for various numbers and combinations of the 16 cylindrical elements. Computed far-field patterns for these various combinations are compared with their respective patterns measured in the far field. (This work was supported by the Bureau of Ships.)

F6. Computation of Far-Field Radiation Patterns from Near-Field Measurements. D. D. BAKER AND C. W. HORTON, *Defense Research Laboratory, The University of Texas, Austin 12, Texas.*—The Helmholtz formula for the wave equation is used to compute the far-field radiation pattern of a large cylindrical transducer from pressure measurements made near the transducer. Radiation patterns are computed for a plane normal to the axis of the transducer by means of a numerical integration over the surface of a circular cylinder of finite length. The normal gradient of the pressure required in the formula is obtained by a simple approximation. The computed patterns compare favorably with measured far-field patterns. (This work is supported by the Bureau of Ships and the Office of Naval Research.)

F7. Transmission and Reflection of a Plane Dilatational Wave at Normal Incidence on a Viscoelastic Layer. G. B. THURSTON AND S. Y. WU (nonmember), *Physics Department, Oklahoma State University, Stillwater, Oklahoma.*—The theory of propagation of a plane dilatational wave is considered for normal incidence on a plane layer. The layer and its adjacent media are assumed to be viscoelastic. Complex transmission and reflection factors are derived for sinusoidal waves. Numerical values of these factors have been computed for propagation through a steel layer and through a butyl rubber layer, these layers separating water and air. The dependence of these factors on the frequency and the thickness of the layer is compared with that for media possessing nondissipative elasticity.

F8. Evaluation af a Boundary Layer Stabilization Coating. J. E. BARGER AND W. A. VONWINKLE, *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—The turbulent boundary layer which forms over submarines at even moderate speeds is found to deter from the vessel's effectiveness in at least two ways: drag increase and noise excitation of the hull plates with attendant increase in the near-field noise interference. Several means exist to minimize the extent of the turbulent boundary layer, although only the distributed damping coating appears amenable to shipboard use. The coating discussed in this paper was of the type first proposed by Dr. M. Kramer of the Coleman-Kramer Company. The coating used was fabricated by the U. S. Rubber Company under the direction of Dr. F. W. Boggs, who has, in addition, presented an improvement to the theory of Dr. Kramer. Two gravity propelled missiles were used as test vehicles; one served as a control and the other was covered with the sample coating. Four very small pressure transducers were mounted at the surface of the missiles along their length. These hydrophones were used to give the details of the boundary layer at their respective locations. Data are presented from which estimates of the effect on transition Reynolds numbers by the coating can be made. Also presented are the pressure coefficient and power spectrum of the boundary layer at various locations on the missile, e.g., various pressure gradients. One parameter of the coating itself is varied to adjust the coating performance to the test Reynolds number.

F9. A Study of the Underwater Noise Produced by a Symmetrical Air Flow Generator. J. J. COOP, *U. S. Naval Air Development Center, Johnsville, Pennsylvania.*—Aircraft such as helicopters, which may hover over water, produce a rather high level of underwater noise. This noise may be produced by engine exhaust, rotor noise, gear noise, or rotor downwash. In an attempt to separate these noise sources, a two-blade rotor having a diameter of about 8 ft was mounted on a 12-×12-ft platform together with a piston motor and an electric motor, as power sources. Provisions were made for three modes of operation, namely: (1) piston engine driven rotor, (2) electric motor driven rotor, and (3) piston engine driving motor as a generator with rotor disconnected and power dissipated into a resistive load. The piston engine power in mode (3) was made equal to that of mode (1). A constant tip speed and pitch angle were used. Measurements were made at the Philadelphia Naval Base, where the water depth was about 40 ft. The rotor unit was suspended over the water by a crane and the rotor operated by remote control. One-third octave band levels of both air and water noise were recorded. From 400 cps to about 2 kc the underwater noise level produced by the rotor exceeded the exhaust level by about 4 db. The results indicated that the underwater noise was produced by airborne noise rather than by rotor downwash.

F10. Dynamics of Stable Cavities during Acoustic Cavitation. H. G. FLYNN, *Acoustic Research Laboratory, Harvard University and Department of Electrical Engineering, University of Rochester, Rochester 20, New York.*—Stable cavities in a field of acoustic cavitation are small gas-filled cavities that have a lifetime which is long compared with the period of the sound field. The oscillations of such cavities in liquids under the influence of a sinusoidal pressure field have been studied to gain insight into mechanisms important in phenomena associated with acoustic cavitation, such as erosion and chemical reactions. A nonlinear differential equation for the motion of the interface of a stable cavity has been solved numerically on a digital computer and analytically by an iterative method. Stable cavities, which may pulsate through many periods of the sound field, may be contrasted with transient cavities whose lifetimes are short compared with the period of the sound field. A transient cavity is a small gas-filled bubble that on expansion to some maximum size contracts violently; during most of the collapse phase a transient cavity acts as though it contained only vapor. The transformation of stable cavities into transient cavities as a function of frequency and pressure will be discussed. (This research is supported by the Office of Naval Research.)

Session G. Architectural Acoustics

Paul S. Veneklasen, *Chairman*

Contributed Papers

G1. Acoustics of Four Large Viennese Halls. E. J. Skudrzyk and E. Hirschwehr (nonmember), *Pennsylvania State University, University Park, Pennsylvania.*—At the Acoustical Society meeting in the fall of 1955, a paper was presented on the development of several types of room sound absorbers. These absorbers, which were tuned to a frequency of about 200 cps, had an absorption coefficient of more than 80% between 200 and 1500 cps. These absorbers have been used in the acoustical design and construction of four large Viennese halls. The largest hall, which is 100 yd long, 120 yd wide, and about 20 yd high, has a reverberation time of four seconds—exactly as planned. As a result of the nearly optimum shape of the hall, the intelligibility for random syllables spoken at the center of the hall is 96% at the remotest seats. This is equivalent to being adjacent to the speaker. Loudspeakers must be used, however, to overcome the noise produced by the audience. The second largest hall is 60 yd long, 30 yd wide, and 20 yd high. The side walls, which are made of glass arranged in a zigzag pattern, do not produce the slightest echo. The end walls are constructed of tuned brick absorbers. From an acoustic standpoint, the skating hall is probably the most impressive. This hall, which is heavily damped particularly at the low frequencies, conveys the impression of a hall in a luxuriant hotel. The only noise that reaches the occupants is the soft scraping of the skates on the ice. This paper presents details of the acoustical construction of these buildings and the results obtained.

G2. Acoustics of the Jerusalem Congress Hall (Binyanae Ha'oomah). Leo L. Beranek and David L. Klepper, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The 3200-seat Congress Hall in Jerusalem, Israel, was completed in the spring of 1960 after a construction period of some ten years. The original primary design goal was to provide a place for large assemblies, the most important of which was the semiannual meeting of the World Zionist Congress. During construction, the repeated use of the partially completed "unroofed" structure for outdoor concerts led to the reconsideration of the original design goals to give primary emphasis to music. The hall now has become the home of the Israeli Philharmonic while still serving its function for congresses and other uses. This paper traces the development of the acoustical design goals as well as discusses the important acoustical design features, the results of measurements made in the completed hall, and the reactions of musicians who have played in the hall.

G3. Acoustical Design of the Jewett Fine Arts Center —Wellesley College, Wellesley, Massachusetts. William J. Cavanaugh, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The music-theater wing of the Mary Cooper Jewett Fine Arts Center at Wellesley College was completed and occupied in the fall of 1958. The building contains a 350-seat auditorium which functions as a music recital hall, lecture hall and, with a portable stage wagon that is moved on tracks from an adjacent drama-workshop, as a small theater. In addition, there are the usual music teaching spaces, including theory classrooms, teaching studios, individual instrumental practice rooms, record-listening rooms and a music library. The room acoustics and sound isolation designs are discussed and the results of some measurements in the completed building are presented. The architects for the building were Paul Rudolph, New Haven, Connecticut, in association with Anderson, Beckwith and Haible of Boston.

G4. Direct and Reverberant Field Statistics in Nearly Hard Rooms. Richard H. Lyon, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Calculations of the first-order density of pressure amplitudes in a reverberant room are reported. The random source is presumed to emit a Poisson sequence of pulses which have random amplitudes but identical shapes. It is shown that sufficiently near the source the distribution of pressure amplitudes reproduces that of the source, and that in the reverberant field there is a tendency to normality due to the great image overlap. The conditions for normality are presented, as well as those for statistical independence of the direct and reverberant fields. An experimental illustration of the theoretical ideas is included. Implications for calculations of random fatigue and malfunction are discussed. (Supported in part by Wright Air Development Division and by B+K Instruments, Inc.)

G5. Frequency Distribution of the Normal Modes of Vibration in Rectangular Rooms. L. W. Sepmeyer, *System Development Corporation, Santa Monica, California.*—Several investigators, notably R. H. Bolt and D. Y. Maa, have examined the statistical distribution of the normal vibration frequencies in rectangular rooms. However, more definitive information would be useful, both for evaluating existing rooms and for optimizing the design of new acoustical laboratories, music rooms or recording studios. Among such information would be more detailed and precise knowledge of the effect of the relative room dimensions on the distribution of its normal modes. An IBM 7090 computer has been programed to determine the frequency distribution of the normal modes of rectangular rooms in each half octave interval over the range of the first four octaves. Frequency has been normalized with respect to the gravest mode, and dimension ratios will be varied from a cube to 5 to 1 parallelepipeds in steps as small as 1%. The principal criterion computed is the rms ratio between the actual mode spacing and the ideal spacing over each half-octave interval. The average mode spacing and, if needed, a more sensitive indicator to help determine the "best" room dimension ratio also will be computed. This paper discusses the equations used, the criteria to be computed, and the results obtained to date.

G6. Angular Distribution of the Normal Modes of Vibration in Rectangular Rooms. L. W. Sepmeyer, *System Development Corporation, Santa Monica, California.*—In rooms used for making sound power measurements as well as for testing acoustical materials diffuse sound fields are postulated. In addition to the requirement for a large number of properly spaced normal modes, good diffusion also requires uniform angular distribution of the modes. Hence, the design and evaluation of reverberant rooms requiring diffuse sound fields require knowledge of the angular distribution of the normal modes as well as their frequency distribution. The angular distribution of the normal modes is determined in one portion of the computer program discussed in the companion paper on the frequency distribution of normal modes in rectangular rooms. This is done by computing the angle each mode makes with the short axis of the room and counting the number which lie within each 10° class interval for each half octave. The criterion used is based on the ratio between the actual number and the ideal number in each class interval. This will be done for the same range of dimension ratios and frequencies used for computing the frequency distributions. The development of the angular distribution criterion will be discussed, and preliminary results will be presented.

G7. Experiments Using a Tilted, Rotating Sample for Absorption Coefficient Measurements in a Reverberation Chamber. J. Ortega and I. Rudnick, *Physics Department, University of California at Los Angeles, Los Angeles 24, California.*—Absorption coefficient measurements in reverberation chambers are particularly troublesome at low frequencies. A major part of the difficulty stems from the fact that the eigenmode density is low and that an abnormally high percentage of the modes are axial and tangential. For a sample mounted on the floor many of these modes have grazing incidence and the associated absorption coefficient is abnormally low. A common manifestation of this effect is the occurrence of double slopes in the decay curve. This situation can be partially corrected by tilting the sample. If further the sample is rotated during the decay, no mode can remain grazing, and the rotation will result in a more appropriate average for the absorption coefficient as a function of angle. A comprehensive set of measurements were made to experimentally determine the size of these effects and the results will be discussed. The outstanding result is that nonlinear decay slopes were not observed when the sample was tilted and rotated.

G8. Optimum Acoustic Criteria of Concert Halls for the Performance of Classical Music. Fritz Winckel, *Case Institute of Technology, Cleveland, Ohio.**—In recent years one has found essential criteria of room acoustics which are responsible for good intelligibility of speech addresses in auditoria. Concert halls which have been built according to these principles are not fully satisfying as to the performance of classical music. The high intelligibility which one deduces from the first reflection pulse and the envelope of the decay curve is not of primary importance for classical music which demands, above all, fullness of tone and blend. An opportunity was afforded to go on tour with the Cleveland Orchestra and to make statistical measurements in 15 different auditoria during their concerts. It has been resulted that the dynamic range attainable by the orchestra is directly dependent on room acoustic qualities. The lower limit of loudness is given by the noise level, variable in the different halls, and the upper limit by diffusion and absorption which are also responsible for the quality or timbre. The behavior of the orchestra is revealed further from the tempo of the music, with the tendency that in the better concert hall with the optimum reverberation time little absorption and good diffusion allow a faster tempo.

* Permanent address: Technische Universitaet, Berlin-Charlottenburg, Germany.

Session H. Psychological Acoustics I

E. E. David, Jr., *Chairman*

Contributed Papers

H1. Calculation of Loudness: 1961 Revision. S. S. Stevens, *Psycho-Acoustic Laboratory, Harvard University, Cambridge 38, Massachusetts.*—Steps have been taken to improve the procedure for calculating the loudness of a complex sound. Two changes have been made. (1) The equal loudness contours for bands of noise in a diffuse field [see Noise Control 3, 11–22 (1957)] have been approximated by straight lines in a log-log plot. (2) The spacing of the contours has been altered to reflect the nonlinear growth that takes place in the loudness level of bands of noise when their width exceeds the critical bandwidth. The basic formula for the addition of loudness across frequency remains the same: the total loudness S_t is given by $S_t=S_m+F(\Sigma S-S_m)$, where S_m is the loudest band and F has the values of 0.3, 0.2, and 0.15 for octave, half-octave, and third-octave bands. The revised procedure has the advantages that it can be described more easily and it agrees better with the available measurements on loudness level.

H2. The Discrimination Limen for Loudness under Varying Rates of Intensity Change. David Wolsk, *Physiological Acoustics Laboratory, University of Michigan, Ann Arbor, Michigan.*—Two studies have been conducted on the influence of the rate of auditory intensity change upon a subject's ability to discriminate that change. In experiment I the stimulus intensity changed gradually through time. Complete psychophysical functions were obtained for four subjects at five rates of change between 2 and 150 db/min. To control for possible differential auditory fatigue effects, a stable level of fatigue was maintained throughout the trials. The slowest rate of change was the only one which produced results significantly different from the other rates and this was so for only two of the four subjects. In experiment II a warbling type of intensity variation was used with four warble rates between 0.3 and 3.0 beats/sec. The rate of change of the warble amplitude was adjusted for each warble rate to serve as a compensation for the fewer number of warble cycles presented in a given observation time with the slower warble rates. This control has apparently not been used before in two previous similar studies whose results indicated a very definite decrement in discrimination as the warble rate decreased. The results from this experiment showed no significant effects on the DL from varying the warble rate. The results will be discussed in terms of individual differences and changing discriminative criteria as they relate to temporal factors.

H3. Loudness Summation and Spectrum Shape. Bertram Scharf, *Department of Psychology, Northeastern University, Boston, Massachusetts.*—The loudness of 3-tone complexes was studied as a function of the intensity relations among the three components. Generally, for a given SPL and a given frequency separation (ΔF) between the two side bands, a 3-tone complex was loudest when its spectrum was flat, i.e., when the components were equally intense. Two types of spectra were investigated: (1) peaked spectra, for which the loudness of the complex was measured as a function of the intensity difference between the middle component and the less intense side bands, and (2) sloped spectra, for which the loudness was measured as a function of the slope of the line spectrum. Although for both peaked and sloped spectra, the loudness of complexes whose ΔF was greater than a critical band decreased as the shape of the spectrum became less and less flat, the decrease in loudness was more rapid for the peaked spectra. The loudness of complexes whose ΔF was less than a critical band changed little as a function of either peakedness or slope. Moreover, the data suggest that reducing the relative intensities of one or both side bands has a similar effect upon loudness as does changing their frequencies to produce a smaller ΔF; both types of change usually produce a reduction in loudness. (This research was supported by a grant from the National Institute of Health, U. S. Public Health Services.)

H4. Pitch of High-Pass Filtered Periodic Pulses. NEWMAN GUTTMAN AND JAMES L. FLANAGAN, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Earlier pitch-matching experiments [J. Acoust. Soc. Am. 32, 1308–1328 (1960)] using patterns of various polarity configurations showed two modes of pitch perception associated with unfiltered stimuli. A third mode appears when the fundamental-frequency component of the pulse train is rejected. The modes appear successively as frequency increases and correlate objectively with pulse rate, fundamental frequency and the lowest frequency component present in the stimulus. By using similar stimuli and procedure, the present study investigates the effects of severe high-pass filtering (2 and 4 kc). Under these conditions, basilar membrane displacement is restricted to medial and basal regions where the impulse response is short compared to the apical end. The subjective response is to prolong the frequency range over which the pulse-rate mode operates. In some cases the fundamental mode does not appear until the fundamental frequency of the stimulus approaches or becomes greater than the cutoff frequency of the filter. In other words, the classical residue (fundamental) pitch associated with the range 200–500 cps may not be heard. As shown in our previous experiments, these subjective phenomena can be related to the displacement response of the basilar membrane. Simulation of displacement is accomplished by electrical analog networks of the middle ear and membrane.

H5. Effect of Masking on the Pitch of Periodic Pulses. AARON E. ROSENBERG, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey, and Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, Massachusetts.*—Pitch-matching experiments were performed with stimuli and procedure similar to those employed by Flanagan and Guttman [J. Acoust. Soc. Am. 32, 1308–1328 (1960)] to determine whether selective masking alters the transition region between the pulse rate and pattern-rate modes of perception. In the unmasked condition, for pulse repetition rates below approximately 150 pps, listeners essentially match the pulse repetition rates of the comparison and standard signals; for fundamental frequencies higher than approximately 150 cps (and below 800 cps), listeners match the fundamental frequencies of the comparison and standard signals, even if the fundamental has been removed from the standard by filtering. Present results indicate that if to standard signals in or near the transition region high-pass noise (1000 cps) is added, the "buzz" quality associated with the pulse rate judgment can be masked and a pattern-rate judgment favored. Conversely, low-pass noise (1000 cps) can mask the "tonal" quality associated with pattern-rate judgments and favor pulse-rate judgments. Equally effective narrow-band noise or sinusoidal maskers seem to lie at about 5000 cps for masking pulse-rate pitch and 500 cps for masking pattern-rate pitch.

H6. Pitch Discrimination for Short Signals as a Function of Interval. JACK C. WHITESELL (nonmember), AND LLOYD A. JEFFRESS, *Defense Research Laboratory and Department of Psychology, The University of Texas, Austin 12, Texas.*—Five subjects adjusted the frequency of one oscillator, the "variable," to match the pitch of a second, the "standard." The stimuli were presented in sequence, first the standard, then the interstimulus interval, then variable followed by a one-second pause before the next presentation. The subject made his adjustments of the frequency of the variable during the one-second pause. The stimuli were presented binaurally by means of PDR-10 earphones. Five frequencies were used for the standard, 300, 600, 1200, 2400, and 4800 cps, and two durations, 25 and 100 msec. The duration of the variable was kept constant at 100 msec. Two interstimulus intervals were used, 100 and 500 msec. The results of the experiment are in agreement with previous work with respect to the effects of frequency and of duration on the threshold, but show rather more dependence on interstimulus interval than was expected. (This work was supported by the Bureau of Ships.)

H7. Pitch Retention by Compensatory Frequency Tracking. J. DONALD HARRIS AND ANDREW G. PIKLER (nonmember), *U. S. Naval Medical Research Laboratory, Groton, Connecticut.*—The accuracy of compensatory tracking of a reference frequency of 250 cps was studied. Over a duration of 6 min this standard was changed at a rate of 3 cycles/second in such a way as to keep each of three programs within the limits of 114 cps on the low, and 370 cps on the high side. One program was symmetrical around 250 cps, one was always the same as or higher than 250 cps, another always the same or lower. Each program contained seven 10-sec plateaus interspersed at 30-sec intervals at 250 cps. Programs and tracking responses were displayed in terms of cps vs time on a paper tape recorder by the device of mounting rotary attenuators on the same shafts as those of the capacitors which allowed for frequency changes. Frequency could be read to the nearest couple of cycles. Analysis of variances showed that, while the 6 trained but generally unmusical *S*'s were somewhat different, the three programs and the three replications of each program did not act as sources of variance. All *S*'s were continuously "undershooting" the programs except on the plateaus. The errors ranged from less than the precision of recording up to 9 cps. It would thus seem that over a 6-min span even unmusical *S*'s can retain rather precisely a pitch sensation in the face of drifts which could, if uncompensated, lead to pitch changes of more than an octave. Evidently the compensated program is sufficient to allow *S* to simulate a rudimentary form of absolute pitch.

H8. Auditory Discrimination of Duration. C. DOUGLAS CREELMAN, *Cooley Electronics Laboratory and Communication Sciences Research Laboratory, University of Michigan, Ann Arbor, Michigan.*—A series of experiments measured human ability to discriminate durations of auditory signals. A two-alternative forced-choice procedure was used: two sine-wave signals of identical amplitude and frequency, differing only in duration, were presented sequentially on each trial. The order of presentation was random, and the task was to state, for each experimental trial, whether the longer signal had occurred first or second. The signals were presented in a background of continuous white masking noise, which was held at a constant level throughout the experiments. In separate experiments the effects on performance of signal voltage, "base" duration, and increment duration were measured. A model was developed within the framework of statistical decision theory to account for the data in terms of a signal-independent "counting mechanism" operating over the relevant durations. Limitations of discrimination are due to uncertainty regarding the end points of the time interval, and memory decay between observations on a single trial. Two parameters were estimated for each observer to fit the data, and the results of two further experiments were predicted. Some implications of the model for psychophysical theory will be discussed. (This research was part of a doctoral dissertation submitted to the University of Michigan. It was conducted under contracts with the Operational Applications Laboratory of the Air Force Command and Control Development Division and the Office of Naval Research.

The author is currently a Postdoctoral Fellow at The Johns Hopkins University, Baltimore, Maryland.)

H9. Binaural Lateralization of Cophasic and Antiphasic Clicks. JAMES L. FLANAGAN, E. E. DAVID, JR., AND B. J. WATSON (nonmember), *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Four subjects listened binaurally to equal-amplitude, 100-μsec rectangular pulses. The pulses were delivered at a sensation level of 40 db and at a rate of 10 sec^{-1}. For wide-band reproduction and good transient response, the stimuli were transduced by condenser microphones fitted with ear insert plugs. The relative times of arrival of the pulses at the two ears (both lead and lag) were under continuous control of the subjects. The latter adjusted the interaural time difference to produce fused sound images. The polarities of the pulses were connected for cophasic (rarefaction-rarefaction) and antiphasic (condensation-rarefaction) conditions. Besides the unmasked conditions, lateralizations were made when the two ears were masked by additive, uncorrelated noise, high-pass filtered at 600, 1200 and 2400 cps, respectively. The results indicate that: (a) the interaural time difference for principal fusion of the cophasic condition, both masked and unmasked, is nominally zero; (b) the interaural time difference for principal fusions of the unmasked, antiphasic condition is on the order of 200 to 300 μsec; (c) masking by uncorrelated noise, high-pass filtered at frequencies less than about 1–2 kc/sec, produces antiphasic lateralizations at time differences of approximately $\pm 1/2f_c$, where f_c is the cutoff frequency of the noise; (d) masking by noise high-passed at frequencies greater than 1–2 kc yields lateralizations essentially the same as the unmasked case. Secondary fusions, generally lying off the medial plane, are frequently heard. Earlier calculations on the mathematical models of basilar membrane displacement [J. Acoust. Soc. Am. 32, 1494(A) (1960)] suggest the present results and, in fact, prompted the experiment.

Session I. Underwater Acoustics—Propagation and Arrays

C. M. MCKINNEY, *Chairman*

Contributed Papers

I1. Air-to-Water Transmission Across a Rough Interface. W. W. SCALES, *Bell Telephone Laboratories, Inc., Whippany, New Jersey.*—The ray solution to the problem of sound transmission across a smooth air-water interface [A. A. Hudimac, J. Acoust. Soc. Am. 29, 916 (1957)] is extended to the case of a rough boundary simulating the ocean surface. A Gaussian distribution of wave slopes was assumed with a variance which is wind dependent. The problem was made two dimensional by considering the wave disturbance limited to swells of indefinite lateral extent. Time averaged transmission losses were computed numerically for a number of source and receiver positions for a point source in air and a receiver in water. A comparison is made for the smooth and rough surface situations. Waves give rise to an improvement in transmission, which increases with wind speed. For a smooth surface the intensity falls off at great ranges as $1/r^4$, while for the rough surface it tends toward $1/r^2$. (This work was supported by the Office of Naval Research under contract.)

I2. A Scaled-Model Experiment of Wave Propagation in a Randomly Inhomogeneous Medium—I. E. O. LACASCE, JR., R. G. STONE (nonmember) AND D. MINTZER, *Laboratory of Marine Physics, Yale University, New Haven 11, Connecticut.*—When a series of uniform pulses is transmitted in the ocean, the received pulses vary about an average amplitude in a random manner. A theory developed by Mintzer [J. Acoust. Soc. Am. 25, 922 (1953)] predicts that $V^2 \sim k^2\alpha^2 ar$, for $kr \gg 1$ and $ka \gg 1$, where V is the coefficient of variation defined as the fractional standard deviation of a series of pulses, $k=2\pi$/acoustic wavelength, α the rms index of refraction variations from unity, a the mean correlation distance, and r the range from source to receiver. In a scaled-model experiment the refractive index variations are caused by heating the medium (water) from below, thus causing turbulent convection. The experimental apparatus will be described and the observations of the dependence of V on the range between source and receiver and on the acoustic frequency will be presented. (Supported by the Office of Naval Research.)

I3. A Scaled-Model Experiment of Wave Propagation in a Randomly Inhomogeneous Medium—II. R. G. STONE (nonmember) AND D. MINTZER, *Laboratory of Marine Physics, Yale University, New Haven 11, Connecticut.*—When a tank of water is heated from below, the turbulent mixing of warmer, more buoyant water produces thermal inhomogeneities and related variations in the index of refraction. The acoustic fluctuations are determined experimentally as a function of the microstructure parameters. The significant parameters of the microstructure are separately measured with thermistors; provision is made to determine the temperature fluctuations at a single point or the correlation of the fluctuations at two points. The relations between the acoustic and the microstructure fluctuations are then determined, and compared with the theory of Mintzer. There is the interesting possibility of determining the parameters of the temperature microstructure by means of acoustic waves. (Supported by the Office of Naval Research.)

I4. Underwater Acoustic Ray Analysis. G. F. GRABER, A. I. RUBIN (nonmember), *Electronic Associates, Inc., Princeton, New Jersey,* AND R. F. SEYMOUR (nonmember), *U. S. Navy Underwater Sound Laboratory, New London, Connecticut.*—A simulation of the trajectory of underwater acoustic rays was implemented on a general purpose electronic analog computer. Equations defining the phenomena are summarized, assumptions are listed and included, and a preferred form of the equations is developed to provide maximum attainable accuracy of solution. Included are considerations of the velocity of sound as an arbitrary function of both depth and range and the effects of reflections from the surface and an arbitrary bottom. Complete computer diagram is presented and special circuits are described. Typical results of depth versus range are shown for varying launch depths and launch angles. A method of obtaining the difference in time required for rays leaving a common emitter at various angles to reach a common receiver is also covered. Finally, provision for computing spreading loss as a function of range is described.

I5. Calculations of Caustic Lines in an Ocean of Constant Velocity Gradient Layers. D. A. MURPHY, *Hughes Aircraft Company, Fullerton, California.*—It is generally recognized that caustics exist in convergence zones near the surface of the ocean. The shapes of these caustic lines along their entire extent from the surface to the bottom of the ocean has been recently studied. Each caustic line consists of two branches connected at a cusp where the caustic first appears caused by a folding wave front. A systematic study of the formation of caustics in media of constant velocity gradient layers has been made. The study starts with the infinite medium with one isovelocity layer above one constant gradient layer and progresses to a many layered ocean closely approximating a real velocity profile and limited by a surface. Calculations of ray paths and wave fronts were made with the IBM 709 computer in order to determine the caustic lines. Plots of range versus initial angle were used to explain points of interest on the caustics. The effects of surface shadow on the caustic lines were shown, and wave fronts in this region were explained. It was found that a two-layer ocean demonstrates the essential features of the caustic lines.

I6. Digital Computer Calculation of Wave Fronts in a Layered Medium of Constant Gradients. E. M. OSTLER (nonmember), *Hughes Aircraft Company, Fullerton, California.*—An IBM computer program has been written to compute equitime surfaces in a medium of constant velocity gradient layers. The ability to generate wave fronts in this manner allows a more thorough investigation to be made of shadow zones and caustic regions, than was formerly possible. It is also now possible to study directly the phase deviations along an array in a medium which closely approximates the real ocean. The usual ray-tracing techniques of geometrical acoustics are used but, in addition to the usual range and transmission time calculations made at interfaces, depth and range coordinates are calculated for prespecified transmission times. These coordinates then define an equitime surface. The equations used in these calculations are simple rearrangements of the equations for range, depth, and transmission time. A constant phase surface can be generated from the equitime surface by introducing appropriate phase shifts at the boundaries.

I7. Maximixing the Directivity of Arrays of Nondirectional Acoustic Receivers. A. H. LUBELL, *Hughes Aircraft Company, Fullerton, California.*—The conditions under which maximum directive gain and receiving directivity are equivalent are reviewed. Then a fundamental method of calculating directivity from given correlation functions for signal and noise based on the original treatise by Faran and Hills is presented. Numerical results are given for line arrays and a particular class of two-dimensional arrays that appear to have optimal realizable directivity of significance in the design of small receiving arrays. Empirical relationships are given for nominal values of directivity and beamwidth of uniform line arrays used for the reception of broadband signals in a background of broadband isotropic noise.

I8. Some Experimental Results Using a Scanned-Line Array to Determine the Direction of a Sound Wave. D. C. WHITMARSH (nonmember) AND M. T. PIGOTT (nonmember), *Ordnance Research Laboratory, The Pennsylvania State University, University Park, Pennsylvania.*—An attempt is being made to employ independent hydrophone elements uniformly spaced along a straight line to deduce the bearing of an incoming plane sound wave. Upon sampling the outputs of the hydrophone elements in succession, one simulates a single hydrophone moving relative to the sound wave. The signal measured in such a case would have a frequency of $f=f_0(1-\alpha\cos\theta)$ where f_0 is the frequency of the sound, α is the ratio of the scanning speed to the sound speed, and θ is the bearing angle to be deduced from a measurement of f. A method to accomplish this was outlined in a paper presented at the June, 1960 meeting of the Society. Since then a hydrophone about four feet long containing 134 independent elements has been obtained and the necessary switching and sampling circuits built to test the method experimentally. Measurements made in a laboratory tank confirm the theoretical results very nicely for a steady signal. These together with the results obtained using a pulsed signal and the effects of noise on the system will also be presented.

I9. Compressional Velocity Determination for the Bottom Layer between Cape May and Montauk Point. PETER A. BARAKOS, *U. S. Navy Underwater Sound Laboratory, Fort Trumbull, New London, Connecticut.*—Pressure waves generated at different ranges on the continental shelf and slope by explosions of one pound of TNT were received by hydrophones connected by sea cables to a fixed station in the ocean. An extensive analysis of the dispersion in the water wave covering five different courses within a 150-mile radius from the receiving station was made with over 400 charges in all the seasons of the year. In general, for shallow water the received pressure wave displays certain features which are characteristic of the depth of the water, the velocity-depth contour in the water, the range, and the structure of the bottom. Dispersion in the water wave was also observed in a surface channel formed in moderately deep water. The observed dispersion phenomena were studied quantitatively and the data were interpreted in terms of the bottom structure. Two and three layer liquid models were used to interpret the experimental results.

I10. Equipment for *In Situ* Measurements of Sound Velocity and Absorption in Sea Floor Sediments. GEORGE SHUMWAY, *U. S. Navy Electronics Laboratory, San Diego 52, California,* AND W. B. HUCKABY (nonmember), *Scientific Service Laboratories, Dallas 21, Texas.*—With the intent of making *in situ* measurements of sound velocity and absorption in deep sea sediments from the bathyscaph *Trieste*, equipment for this purpose, useful also in shallow water, has been developed and field tested. Three probes, each containing two transducers 0.5 m apart, are fastened to a rigid beam for simultaneous insertion into the sediment. The receiving probes lie 1 and 2 m distant from the source probe. Pulse-actuated magnetostrictive transducers are used as sound sources, and barium titanate cylinders as receivers. Measurements at two depths in the sediment are possible by using upper or lower transducers. The travel time interval between reception of the transmitted pulse at the near and far receivers is measured on a decade scaler counting a precision oscillator. Oscilloscope presentation of the signals arriving at the two receivers along with the time of gating assures the operator of satisfactory operation and aids in setting certain controls. A frequency range between 7 kc/sec and 30 kc/sec is used, with filtering provided in the receiving circuit to allow measurements in a number of frequency bands. Absorption measurements are made by matching superimposed oscilloscope traces of the signals from the near and far receivers by means of precision bridge T attenuators in two matched amplifier channels.

Session J. Speech Communications II

WEIANT WATHEN-DUNN, *Chairman*

Contributed Papers

J1. Electromyography as a Speech Research Technique with an Application to Labial Stops. GLORIA LYSAUGHT (nonmember), ROBERT J. ROSOV (nonmember), AND KATHERINE SAFFORD HARRIS, *Haskins Laboratories, New York 17, New York.*—Electromyographic techniques have been developed as indicators of the physiological processes of speech production. They provide a sensitive and direct means for describing the speech gesture in terms of the activity of the actuators rather than the consequent displacements of the speech mechanism. To the extent that perception is geared to articulation, this *action pattern* may be expected to yield simpler descriptions of the basic units of speech than have been available in acoustic or traditional phonetic terms (Franklin S. Cooper, Alvin M. Liberman, Katherine Safford Harris, and Patti Murray Grubb, "Some input-output relations observed in experiments on the perception of speech," *Proceedings of the Second International Congress on Cybernetics,* Namur, 1958). This account attempts no more than a description of the experimental techniques and their application to a specific study of the English labial stops /p,b/. The electromyographic measures showed no significant differences in timing or amount of muscle activity at the lips between /p/ and /b/. In this pilot study, the electromyographic method has proved feasible and has yielded a partial description of /p,b/ that is simply structured and consistent with the acoustic-perceptual characterization of these sounds.

J2. Identification and Discrimination of a Phonemic Contrast Induced by Silent Interval. JARVIS BASTIAN (nonmember), PETER D. EIMAS (nonmember),* AND ALVIN M. LIBERMAN,* *Haskins Laboratories, New York 17, New York.*—At a previous meeting of this Society we described research which indicated that the insertion of silent intervals into recordings of natural productions of certain fricative-vowel sequences is sufficient to induce the perception of stops. For example, introducing a sufficiently long silent interval after /s/ in a production of *sore* causes the listener to hear *store.* Further research is presented on the perception of such intervals in the contrast between *slit* and *split.* Stimuli containing various durations of silent interval were presented to subjects for (phonemic) identification as *slit* or *split.* The same stimuli were presented also for forced-choice discrimination on the basis of any differences the subjects could hear. A comparison of the results indicated that discrimination is most acute in the region of the phoneme boundary. Analysis of the data showed that accuracy of discrimination was determined almost entirely by whether or not the sounds in a given comparison were perceived as different phonemes. Thus, the perception of this acoustic continuum can be regarded as essentially categorical. These results are comparable to those obtained in other studies of consonant perception. The categorical perception of the consonants may be explicable in terms of the categorical nature of their articulatory gestures.

* Also University of Connecticut.

J3. Mimicry and the Perception of a Phonemic Contrast Induced by Silent Interval: Electromyographic and Acoustic Measures. KATHERINE SAFFORD HARRIS, JARVIS BASTIAN (nonmember), AND ALVIN M. LIBERMAN,* *Haskins Laboratories, New York 17, New York.*—In the preceding paper of this session (J2) it was found that a phoneme boundary divides a continuum of silent intervals into two perceptual categories. The purpose of the present experiment was to see if the subjects' mimicry of the same acoustic continuum is graded, or whether it is, like the perception, categorical. Acoustic and electromyographic recordings were made of the productions of subjects during mimicry of each of the above stimuli. The acoustic records showed that time intervals introduced in mimicry were indeed categorical; however, the possibility remained that the speakers were, in some sense, making tentative or partial p gestures in the middle range of the continuum. Electromyographic recordings were used to test for the subjects' tendency to make such gestures. The electromyographic results were also entirely categorical: Either there was a normal burst of muscle potential at the lip (indicating a p gesture) or there was not. Thus, there was no evidence that subjects produce intermediate gestures, such as partial closure, in response to stimuli near the center of the continuum.

* Also University of Connecticut.

J4. Identification and Discrimination of Phonemic Tones. ARTHUR S. ABRAMSON, *Haskins Laboratories, New York 17, New York.*—Measurements of the fundamental frequency patterns of many sets of tonally differentiated words yielded average curves for the five phonemic tones of Standard Thai (Siamese) on both short and long vowels. Experiments were then run to determine the role of fundamental frequency patterns in the perception of these tones. There were three major results: (1) Thai subjects easily identified tones in isolated monosyllables. (2) Highly intelligible tones could be synthesized on the Haskins Laboratories intonator with the average tonal contours that had emerged from the measurements. (3) When the five frequency contours were imposed synthetically on each of five utterances minimally distinguished by tone, the pitch contours overrode the effects of all other features observed in the stimuli. The overriding role of these contours opens the way to the study of discrimination at phoneme boundaries in the domain of prosodic features. The mid and high tones of Thai were chosen as a suitable pair for a start in this direction. These experiments had two results: (1) Shape difference is a stronger cue than pitch height for phonemic identification. (2) The hypothesis that there will be little or no sharpening of discrimination at this kind of phoneme boundary was supported or at least not contradicted.

J5. Effects of Vowel Context on the Articulation of Stop Consonants. OSAMU FUJIMURA, *Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge 39, Massachusetts, and University of Electro-Communications, Chofu, Tokyo, Japan.*—Movements of the lips and the mandible were studied by means of a stroboscopic motion picture techinque during the production of intervocalic velar (palatal) and alveolar stop consonants. Three subjects pronounced English words like *veto, echo, cookie, okay,* etc., in which significantly different labial articulations are required for the first vowel and the second vowel. A film speed of 60 frames/sec was employed for the experiment. Results indicate that during the closure period of nonlabial stops, the labial configuration changes considerably in anticipation of the following vowel. In *echo,* for example, the lips start to close during the production of the first vowel, and the most conspicuous change in their shape occurs during the closure period of /k/. In general, there is substantial disagreement between the movement of the lips and that of the mandible. Some details of the relation between the articulatory movement and the boundaries found in the sound spectrogram will be given. Implications concerning the acoustic structures of the output sound, such as the "target" of formant transition, will be also discussed. [This work was supported in part by the U. S. Army (Signal Corps); the U. S. Air Force (Office of Scientific Research, Air Research, and Development Command); the U. S. Navy (Office of Naval Research); and the National Science Foundation.]

J6. A Study of /w/ and /y/ in American English. ILSE LEHISTE, *The University of Michigan, Ann Arbor, Michigan.*—The study reported in this paper deals with the acoustic manifestations of various allophones of /w/ and /y/ in American English. Occurrences of initial /w/ and /y/, followed by different syllable nuclei, are described. The manifestations of initial /w/ and /y/ are compared with the syllable nuclei /u/, /ʊ/, /i/, and /ɪ/. In addition, the second components of the glides /eᴵ/ and /oᵁ/, and of the diphthongs /ɑʊ/, /ɑɪ/, and /ɔɪ/ are considered. The characteristics of initial /w/ and /y/ include both steady state cues and transitional cues. The second component of the diphthongs /ɑɪ/ and /ɔɪ/ falls approximately in the range of /ɪ/ as a syllable nucleus; the second component of the diphthong /ɑʊ/ has no obvious overlap with the ranges of either /u/ or /ʊ/. No immediate identification of initial /w/ and /y/ with the second components of glides or diphthongs appears possible.

J7. Methods of Measuring Speech Formant Bandwidths. H. K. DUNN, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Accurate measurements of the human vocal tract damping, or the associated formant bandwidths, are not easy. Several different methods have been used by experimenters, with rather widely divergent results. Some of these methods are examined for sources of error, and the conclusions are checked by applying the same methods to the electrical vocal tract. This analog of the human tract can be held constant during a test, can be adjusted in damping, and can be measured accurately by single frequencies. More exact measurement of the real tract would seem to require an artificial source, applied in the pharyngeal cavity, having a lower fundamental and a more accurately known spectrum than the vocal cord tone. A few measurements are reported, obtained by means of an artificial larynx at 30 pulses/sec, applied to the outside of the throat wall. Analysis is made from a tape loop, by means of a narrow band wave analyzer. The results are not wholly satisfactory because the spectrum inside the tract is not accurately known.

J8. An Accurate Estimate of the Glottal Waveshape. M. V. MATHEWS, J. E. MILLER (nonmember), AND E. E. DAVID, JR., *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—The volume velocity function of the glottis can be computed from a pitch-synchronous spectral analysis of the speech pressure wave. The method consists of computing the Fourier coefficients of a pitch period, locating the formant frequencies, removing the formant poles from the spectrum, and regenerating the glottal waveform from the residual by Fourier synthesis. To obtain accurate results it is necessary to record the original speech without phase distortion while maintaining at least 55–60 db signal-to-noise ratio. The data reported here were derived from recordings originating with a WE-640-AA condenser microphone in a free-space room feeding a digital tape recorder. Two male speakers uttered six vowel sounds between h and d at different pitches and intensities. The glottal waveforms were examined to determine both shape and phase with respect to the speech wave as a function of the sound, the pitch and the intensity. Sequences of periods were studied for variations from period to period.

J9. A Laminagraphic Study of Vocal Intensity. HARRY HOLLIEN, *University of Wichita and Institute of Logopedics, Wichita 14, Kansas,* AND JAMES F. CURTIS, *Department of Speech Pathology and Audiology, State University of Iowa, Iowa City, Iowa.*—It has been suggested that vocal fold cross-sectional area (or mass) varies as a function of vocal intensity. Since this notion has not been subjected to test, it was the purpose of this investigation to study the relationship of the coronal cross-sectional area and mean thickness of the vocal folds to variation in vocal intensity. Subjects included eight males chosen on the bases of age, absence of voice problems, and the ability to produce specified vocal tones easily. The range of fundamental frequencies that each subject could produce was determined by a standard procedure. Final selection was made on the basis of these data and all subjects had essentially the same pitch ranges. Subjects were required to produce each of three vocal pitches at three vocal intensities. Pitches were specified in relation to total pitch range and were located as proportions above the lowest sustainable tone. The 10%, 25%, and 50% points to the nearest semi-tone were used but since all subjects' ranges were similar, frequencies of 104, 139, and 233 cps represented these levels. Intensities were specified as the 15%, 50%, and 85% levels of each subject's vocal intensity range. These proportions were obtained by requesting each subject to produce the loudest and softest phonations possible at each of the three frequencies. Vocal intensity was controlled by an Altex M40 microphone, a B and K calibrated amplifier, and SUI Model 11 sound level monitor. This system was calibrated by the GR 759-B sound level meter which was originally used to obtain the intensity ranges. Laminagrams were made of subjects phonating each of these vocal pitch-intensity combinations and measurements made of vocal fold cross-sectional area and mean thickness. Statistical analysis was carried out by an AXBXS three dimensional analysis of variance design on a IBM 610 computer. Briefly, the results permit the following conclusions: (1) As expected, vocal fold cross-sectional area and mean thickness were systematically reduced with increases in fundamental frequency of phonation. (2) Since the F for area was not significant at the 5% level but the F for mean thickness was, there is some question concerning whether there are systematic changes in vocal fold cross-sectional area as vocal intensity is varied. This question and limitations of the study will be discussed. (This research was supported by a grant from the National Institute of Health.)

Session K. Sonics and Ultrasonics—Bioacoustics

FLOYD DUNN, *Chairman*

Invited Papers

K1. Effects of Low-Frequency Vibrations on Man. HENNING E. VON GIERKE, *Aerospace Medical Laboratory, Wright-Patterson Air Force Base, Ohio.*—The physical responses of the human body to frequencies below approximately 50 cps can be described by analyzing the body as a lumped parameter system. Recent refinements of measurement techniques allow more precise determination of this mechanical model and of many of its parameters, its resonances and mechanical impedances. This model is useful in studying the dynamics of the body exposed to various types of dynamic loads: whole body vibration, transient acceleration, blast exposure, respirator excitation, and rapid decompression. The response of the thorax-abdomen system is discussed in detail. These studies are valuable as guides in future experimentation, in the interpretation of various types of biological effects, and in developing and understanding protective measures. The limits of linearity of the various components of the system and data on tissue strength are still unknown. The various effects limiting voluntary human tolerance to vibrations and their connection with the body's physical responses are reviewed. The inadequacy of existing data for determining definitive vibration exposure criteria for practical application is discussed.

K2. Ultrasonic Modification of Human Brain Structures for Treatment of Neurological Disorders. WILLIAM J. FRY, *Biophysical Research Laboratory, University of Illinois.*—The methodology of preparation and irradiation by focused ultrasound of brain structures in human patients to favorably modify symptoms of various neurological disorders will be described. This includes the method of brain landmarks location and subsequent geometric positioning of the focus of the ultrasonic beam in arrays of sites in specific structures. The determination of values for the irradiation parameters appropriate for producing selective changes in the tissue components, including the control of auxiliary physical variables which specify the state of the tissue during exposure, will be discussed. Design characteristics of the irradiation, calibration, positioning, and control instrumentation necessary to achieve optimum effects at the present time will be outlined. The techniques and results of irradiation will be illustrated by a motion picture showing human patients undergoing treatment. The modification of abnormal involuntary movements during successive ultrasonic treatment sequences in individuals suffering from Parkinsonism and dystonia musculorum deformans will constitute the clinical material of the motion picture.

K3. Ultrasonic Techniques in Ophthalmology. GILBERT BAUM (nonmember), *Albert Einstein College of Medicine and Bronx Veterans Administration Hospital, Bronx, New York,* AND IVAN GREENWOOD (nonmember), *General Precision Inc., Pleasantville, New York.*—Ultrasonic visualization is clinically important in ophthalmology because it makes possible accurate diagnosis of conditions existing in the soft tissues of the orbit and the light-opaque eye. No other diagnositic method short of surgery can yield comparable information. Short-pulse, focused-beam, 15-Mc apparatus is used. Serial distance-azimuth scans are displayed on a radar PPI and are photographed. Inherent resolution is better than 0.2 mm. Technique topics to be discussed include: high resolution, compound scanning, quantitative reflection coefficient measurement by a photo-densitometric method, artifact identification, and three-dimensional ultrasonograms. Illustrative clinical ultrasonograms and their interpretations will be presented.

Contributed Papers

K4. Recording of Heart Wall Motion with Ultrasound. JOHN M. REID (nonmember), *Moore School of Electrical Engineering, Electromedical Division, University of Pennsylvania, Philadelphia 4, Pennsylvania.*—An ultrasonic echo-ranging apparatus using a pulse transmitter and high gain broadband receiver connected to a barium titanate transducer has been used to obtain echoes from the walls and septa of the beating heart. The apparatus can be used over a center frequency range of 0.5 to 2.5 Mc. The sound beam is directed into the heart through the spaces between the ribs, and has been applied directly to exposed hearts of dogs. Curves representing the distance of the walls from the transducer can be recorded during the cardiac cycle. The interpretation of records obtained using frequencies of 1 and 2 Mc on dogs and human volunteers is discussed.

K5. Effect of Ultrasound on Osmotic Fragility of Canine Erythrocytes. EUGENE ACKERMAN, *Section of Biophysics, Mayo Clinic, Rochester, Minnesota.*—Dilute suspensions of canine erythrocytes were exposed to 1 Mc focused ultrasonic fields with maximum acoustic pressures estimated at 10 atm. In standing wave patterns in which circulation was inhibited, the cells clumped rapidly at the nodes and no changes were found. With a free surface, fountaining and stirring occurred. Exposure at 1 atm ambient pressure then resulted in rapid lysis. Erythrocytes exposed briefly in isotonic Rous-Turner mixture and controls were both diluted with sodium chloride solutions of varying concentration. The optical density at 650 mμ was plotted as a function of salt concentration. The exposed samples showed a greater fractional decrease in optical density at low sodium chloride concentrations than did the controls. Other samples exposed in hypotonic solutions showed both the greater sensitivity to low sodium chloride concentrations and a smaller change in optical density as the sodium chloride was varied from isotonic to 70% isotonic. It has not been possible to demonstrate that any factor other than cavitation was responsible for the observed changes in osmotic fragility.

K6. Cellular Changes Induced by Intense Ultrasound. JOSEPH C. CURTIS (nonmember), *Biology Department, Brown University, Providence 12, Rhode Island.*—An extensive study has been made of the cellular changes occurring in mouse liver as a result of exposure to intense, focused ultrasound. In this study a broad range of doses of continuous and pulsed ultrasound have been employed. Groups of animals have been sacrificed at different times after irradiation and the exposed tissue examined, a number of cytological and histochemical techniques being used. Evidence of a selective action of pulsed ultrasound has been found both immediately after irradiation and at later times. This localized action is related to the histological organization of the liver and appears to selectively affect the parenchymal cells. Data on the relation of the amount of hepatic injury produced by ultrasound to several dosage parameters will be presented. The observed effects of ultrasonic action will be discussed in terms of the hepatic architecture and cell structure.

K7. Ultrasonic Absorption and Reflection by Lung Tissue. Floyd Dunn and William J. Fry, *Biophysical Research Laboratory, University of Illinois, Urbana, Illinois.*—The acoustic reflection and absorption coefficients of both normal and diseased (pneumonitis) excised lung tissue (dog) were experimentally determined at a frequency of 0.98 Mc/sec. It is found that the physiological saline-lung interface reflects 50% of the sound energy impinging at normal incidence. The acoustic amplitude absorption coefficient per unit path length of lung tissue is 4.7 cm^{-1}. The very high absorption exhibited can be explained as the result of radiation of acoustic energy by the pulsating gaseous structures of the lung tissue. The theory indicates that the absorption coefficient of lung tissue should approach a minimum as the frequency is increased above 1 Mc/sec and should then increase at still higher frequencies. The diseased lung exhibited an acoustic absorption coefficient approximately 25% less than that of normal lung specimens. (This work was aided by a grant from the American Thoracic Society.)

Session L. Noise

Lewis S. Goodfriend, *Chairman*

Invited Paper

L1. Noise Considerations in the Design and Operation of the Supersonic Transport. Harvey H. Hubbard and Domenic J. Maglieri (nonmember), *National Aeronautics and Space Administration, Langley Research Center, Langley Field, Virginia.*—The main sources of noise for the supersonic transport are noted to be power plants, aerodynamic boundary layers, and shock waves. Discussions are given on the state of knowledge of these noise environments and the manner in which noise considerations may affect the design and operation of this type of aircraft. Engine noise problems related to aircraft structural damage, exposure of ground crews, and reactions of communities near airports are discussed for various proposed power plants including the turbofan. Boundary layer noise considerations include possible fatigue damage to external skin surfaces at high dynamic pressures as well as the internal noise environments which may affect voice communications and passenger and crew comfort. Sonic boom considerations include the generation and propagation as well as the associated effects on structures, equipment, and personnel. Special attention will be devoted to discussions of operating procedures useful for minimizing sonic boom disturbances on the ground.

Contributed Papers

L2. Noise Generated by Aircraft in Flight. Kenneth McK. Eldred, *Western Electro-Acoustic Laboratory, Inc., Los Angeles 64, California,* Thomas R. Rooney (nonmember), and Richard F. Carmichael (nonmember), *Northrop Corporation.*—The generation of noise from turbulent boundary layers has been investigated both theoretically and by model experiments in recent years. However, until the present, no flight data have been available with the exception of high speed flybys. The paper presents results of several inflight measurements of aircraft noise including jet noise, wake noise, and a radiated boundary layer noise, obtained by flush microphones mounted in laminarized pods external to the aircraft. The data indicate that the radiated boundary layer noise is primarily of quadrapole origin at flight Mach numbers above 0.5, and further, that the inflight measurements are essentially in accordance with the ground flyby measurements when the effect of motion on the transmission of sound is considered.

L3. Missile Flight Sound Levels—Telemetering Techniques and Interpretation of Data. J. J. Van Houten, *Convair Division of General Dynamics, Pomona, California.*—Increased emphasis concerning the effects of noise on missile performance has resulted in measurements of flight sound levels utilizing various techniques. In this paper the application of broad band FM telemetering is shown to provide reliable flight measurements. Representative data acquired by use of a condenser microphone system which directly modulated a FM telemetering transmitter is illustrated. Narrow band power spectrum analysis is used to resolve the acoustic signal from system electrical noise. This technique is demonstrated by use of specific examples of telemetered data. Also shown is the correlation of sound level fluctuations and missile maneuvers. It is theorized that these fluctuations result from attached shock wave relocation. Finally, periodic fluctuations in the measured signal strength becomes shown to exist when telemetered signal strength is marginal. These fluctuations result from interference cancellation of the signal when there are two paths to the receiver antenna—one direct path and one reflected off the ground.

L4. Far-Field Noise Effects from Static Tests of the Saturn Booster. W. D. Dorland, *Huntsville, Alabama.*—The acoustic noise generated by the Saturn booster is characterized by the extremely large powers, broad directivity, low frequency power spectra, and low efficiency. Results obtained from a field survey of sound from the Saturn's initial series of static tests reveal some unique aspects of sound generated by clustered jet streams. Initial firings were of two and four engines each which produced acoustic powers of 0.56 and 1.6 megawatts, respectively. Corresponding acousthe efficiencies were quite low, being 0.03% for the two-engine test and 0.04% for the four-engine test. However, acoustic power radiated from eight-engine tests ranged from 25 to 40 megawatts, with an acoustic efficiency near 0.5%. Spectra from all tests peaked between 10 and 100 cps, and were rather ragged, showing severe dips near 250 cps and minor peaks at 1000 and 6300 cps. Directivity was quite broad in the eight-engine tests and much sharper in the two- and four-engine firings. While it is desirable to isolate cluster effects on noise generation it is difficult to do so because of the complicated jet deflector used on the tests.

L5. Sonic Block Silencing for Axial and Screw Type Compressors. J. A. Sobel III (nonmember), and A. D. Welliver (nonmember), *Curtiss-Wright Corporation, Que-*

hanna, Pennsylvania.—The sonic block concept has proven to be an extremely effective method of reducing compressor radiated noise. This type of silencer is based upon the classical fact that when a fluid is traveling in a direction opposite to and at a speed equal or greater than the velocity of propagation of a disturbance in that fluid, then the disturbance cannot be transmitted upstream through the fluid. Since the principle of operation involves only direction and relative speeds, its acoustical performance is independent of frequency. The amount of noise reduction is limited only by (1) suppressor wall noise radiation; (2) noise escape through boundary layer at the sonic block; and (3) fluid acceleration noise. This paper describes sonic block designs for turbojet engine axial compressors and small industrial screw type units. Actual noise reductions are presented for a 225 lb/sec axial flow compressor and a 2 lb/sec Lysholm type unit. In the case of the high speed axial flow unit, high frequency blade noise was submerged below the levels of background noise. Noise reductions, ranging from 28–48 db, were measured from 20–10 000 cps for the screw compressor.

L6. Application of a Blowdown Wind Tunnel for Large Scale Acoustic Environmental Testing. William H. Mayes (nonmember), and Philip M. Edge, Jr. *National Aeronautics and Space Administration, Langley Research Center, Langley Field, Virginia.*—A unique application of a blowdown wind tunnel for intense noise testing of large components or of entire vehicles will be described. The above facility, which is primarily used for thermal structures test, ing, has a 12-ft diam exhaust jet which generates an extensive noise field useful for test purposes. Sound pressure levels in excess of 140 db exist over distances of about 100 ft in the radial direction and 200 ft in the axial direction. Sound pressure levels up to 160 db and a wide range of noise spectra are available. An example of the use of this facility in the environmental testing of a manned space vehicle is cited and some of the main findings are presented.

L7. Model Studies of Some Methods of Jet Noise Generation for Laboratory Testing. C. Thomas Modlin, Jr. (nonmember), and Philip M. Edge, Jr., *National Aeronautics and Space Administration, Langley Research Center, Langley Field, Virginia.*—A description will be given of some studies relating to means of increasing the noise generated by jets to enhance their use as laboratory noise sources in research and environmental testing. A series of special nozzles having two-in. exit diam. and incorporating several types of upstream bends were investigated over a range of temperatures up to 700°F. Over-all sound pressure levels and one-third octave band spectra were measured in the near and far noise fields. Compared to conventional converging nozzles, those incorporating upstream bends resulted in increases of up to 15 db in the over-all sound pressure levels in the near field. Some further increases were obtained as a result of increasing the air temperature. Significant changes in the spectra were also noted and will be described.

L8. Use of Helium and Steam in Acoustic Scale Model Jets. Walter V. Morgan, *Aero-Space Division, Boeing Airplane Company, Seattle 24, Washington.*—It is postulated that design of a valid acoustic model to simulate jet engine exhaust need consider only three exhaust gas parameters: Mach number, density, and velocity. This provides a logical basis for a simplified model testing technique which uses a convenient working medium other than that of the full-scale engine. Operation at reduced temperatures is the major benefit. The design of experiments using steam and helium as the working media to simulate military and afterburning conditions of a jet engine, respectively, is described. The experimental results of these tests and corresponding tests performed with heated air jets support the postulation. Conclusions concerning future applications of this technique are discussed. (This work has been supported by U. S. Air Force contract.)

L9. Noise Rating Method for Fluorescent Lamp Ballast Transformers. William C. Sperry, *Physics Division, Armour Research Foundation, Chicago 16, Illinois.*—The noise radiated from a composite fluorescent lamp system (ballasts, lamps, fixtures, and supporting structure) is dependent to some degree upon the physical and geometrical characteristics of all components. Analytical representation of the composite continuum is inhibited by many difficulties, not the least of which is the wide variation in the various composite systems that can occur. However, the mechanical impedance concept presents a powerful tool for systems of this type where selected measurements can be performed to greatly reduce the analytical effort. Theory is presented to analytically represent the composite system in terms of mechanical impedance. Measurements utilizing a force sensor were made for a number of ballasts which yield information useful in the general theory. A rating method was devised for the ballasts, independent of the rest of the system, which compares their noise-generating capabilities. The method is based upon the measured results of over-all force levels increasing monotonically with input electrical power. (This research was sponsored by Sola Electric Company.)

L10. Noise Characteristics of Several Types of Cooling Tower Installations. Robert M. Hoover, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Results of noise measurements of five cooling tower installations are presented. Included are data on: (1) both induced and cross draft flow towers using indirect drive, slow speed propeller fans; (2) a forced draft tower using centrifugal fans; and (3) an induced draft tower with a direct drive, medium speed propeller fan. These data include both measurements of fan and water flow noise at close-in and distant positions. Also, as two of the tower installations were enclosed in office building penthouses, some of the data can be used as a basis for measures aimed at controlling the transmission of airborne noise from the cooling tower penthouse to occupied spaces in the building. Noise control measures employed at these cooling tower installations are briefly discussed, and where possible the data are related to previously published procedures for estimating cooling tower noise spectra and levels.

L11. Sound Power Measurements on a Competitive Group of Current Room Air Conditioners. Warren E. Blazier, Jr., *York Division of Borg Warner Corporation, York, Pennsylvania.*—From a practical viewpoint, engineering of "quiet" products must be tempered by the market requirements of size, function and price. It is necessary, therefore, to know something about typical sound levels of particular product types in order to select an objective design criterion for noise control which is not unduly conservative. Such a study has been made on room air conditioners, of a nominal 1-hp capacity, by sampling a total of 12 models from four major manufacturers. Data are presented in terms of the sound power envelope (range of maximum to minimum levels) and presumably represent the current state of art in controlling noise on products of this type. Assuming that at least some of these units have sound levels which are nominally acceptable to the customer, it is surprising to note that a design below NC-50 (relative to a quiet bedroom) is conservative by present standards. The range of data on measurements of the "outside" noise of these units is also included. A form of nomogram is presented for evaluating these results in terms of the non-air-conditioned neighbor as well as pending municipal noise codes.

L12. On the Problem of Sound Reduction in Vehicles. C. BETZOLD (nonmember), *A. Stankiewicz Company, Celle, West Germany.*—The development of modern rail and road vehicles has been accompanied by greater attention to acoustical considerations, particularly as they contribute to travel comfort as well as safety. In general, there is strict separation of the four applications: (a) airborne sound absorption; (b) structure-borne sound absorption; (c) structural sound isolation; and (d) airborne sound isolation. It is recommendable that greater use be made of the combination of these techniques in order to attain worthwhile reduction of the sound levels in vehicles. Synthetic materials suitable for airborne isolation and structure-borne absorption have been on the market for several years. In addition to the mentioned acoustical-physical requirements, there are also special technical requirements related to vehicle designing. Such may be light weight, resistance to gasoline, diesel oil, and weather; durability, mechanical strength, etc. Within the range of problems of vehicular acoustics, the special problems of airborne noise isolation by means of special synthetic materials will be treated.

Session M. Geoacoustics

LEON KNOPOFF, *Chairman*

Invited Papers

M1. Speed of Sound Structure of the Atmosphere. WILLIS WEBB (nonmember), *White Sands Missile Range, White Sands, New Mexico.*

M2. Quantitative Ray Theory in Seismology. J. F. GILBERT, *Geophysical Services Inc., Dallas, Texas.*

M3. Recent Studies on Body Waves in the Mantle of the Earth. I. LEHMAN, *Geodetic Institute, Copenhagen, Denmark.*

M4. Inversion of Surface Wave Dispersion. L. KNOPOFF, *Institute of Geophysics, University of California at Los Angeles, California.*

M5. Observations of Long Period Surfaces. J. OLIVER, *Lamont Geological Observatory, Columbia University, Palisades, New York.*

M6. Comparison of Theory and Observation on the Free Oscillations of the Earth. N. NESS (nonmember), *Institute of Geophysics, University of California, Los Angeles, California, and National Aeronautics and Space Administration, Silver Spring, Maryland.*

Contributed Papers

M7. High-Altitude Acoustic Research. JOHN W. WESCOTT, *Institute of Science and Technology, The University of Michigan, Ann Arbor, Michigan.*—The sources and characteristics of sounds at altitudes of 60 000 ft were investigated with a balloon-borne acoustic probe and telemetering system. The system is described and samples of analyzed data presented. Wideband background levels of 0.2 d/cm^2 were measured consistently. Spectral energy per cycle was found to decrease with frequency by about 6 db per octave, a result in good agreement with current theories of noise radiated by turbulence. Cross-correlated data from a two-probe array indicated that most of this energy propagates from lower altitudes. It follows that a major source of high-altitude acoustic noise is lower altitude atmospheric turbulence. In addition to background noise, some specific types of acoustic signals have been detected and identified. Analyzed samples are presented. Some observed Doppler effects are discussed and their possible use for calculating absorption coefficients of low frequency sound in air is explained. (The acoustic probe and financial support for this work were furnished by the U. S. Army Signal Missile Support Agency under contract.)

M8. Instrumentation for Performing Cross-Correlation of Acoustic and Seismic Data on an Analog Computer. JAMES H. PROUT AND RAYMOND E. CRABTREE (nonmember), *Institute of Science and Technology, The University of Michigan, Ann Arbor, Michigan.*—In an experiment designed to study acoustic background noise at high altitudes, a vertical array of two microphones spaced 300 ft apart was hung from a balloon. To determine whether the noise was propagating upward from the jet stream or whether it was caused by local hydrodynamic flow, the signals from the two probes were cross-correlated on an analog computer. The PACE analog computer was chosen for the application because of its ability to handle frequencies as high as 150 cps. An adjustable time delay between channels was provided by a special magnetic tape transport with movable playback heads. Frequency analysis of seismic signals from two large explosions was also performed on the computer by cross-correlating the data with artificially generated sinusoids of various frequencies. Because of the nature of the seismic signal, it is necessary to use special circuitry to permit the accurate measurement of the amplitude of any specified frequency component for any portion of the signal, thus giving an accurate indication of time of occurrence of this particular frequency. Instrumentation for each of these tests is explained and some results are shown for each case. The results of the frequency analysis of the seismic signal are compared with those obtained by conventional filtering methods.

Session N. Accelerometry

D. C. KENNARD, *Chairman*

Invited Paper

N1. Calibration Techniques Related to Vibration Pickup Use. S. EDELMAN, E. JONES, AND E. R. SMITH (nonmember), *National Bureau of Standards, Washington 25, D. C.*—Review of recent publications indicates that current primary laboratory-calibration methods provide sufficient accuracy for present and foreseeable future needs. An upper frequency limit for vibration measurement was first indicated by Schloss. For usual levels, this sets a lower bound to the amplitudes needed in calibration. The use of resonant systems provides means for reaching large amplitudes as they are needed. Serious errors in calibration result from transverse shake table motion. Methods for monitoring motion exist, allowing elimination of this source of error. Shakers free of transverse motion are needed urgently. Development of means for accurate measurement of the transverse sensitivity of pickups at all frequencies of interest is needed also. Full exploitation of primary calibration techniques requires suitable comparison shakers and transfer standards. Available instruments are described. The need for an adequate transfer standard pickup and the characteristics of such an instrument are discussed. To detect damage in storage and attachment or loss of accuracy caused by structural conditions or in associated circuitry, pre-use calibration checks can be performed in a number of ways. Most available methods separate the pickup from the structure and are limited in frequency and amplitude. A new instrument described by Orlacchio and Schilling allows in-place checking of entire systems over wide ranges of frequency and amplitude. Some shortcomings of the method have been indicated by Pennington and alternatives have been suggested. Published advantages, objections, and inadequacies of each method are discussed. Also, we suggest comparison of the attached pickup with a detachable, laboratory-calibrated, transfer standard, both driven in place by a pulse.

Contributed Papers

N2. Added Reliability for Preflight and Inflight Acceleration Measurements. A. W. ORLACCHIO (nonmember), *Gulton Industries, Inc., Metuchen, New Jersey.*—It has become essential to the proper functioning of the missile that a positive means for checking the accelerometer after installation be made available. A newly developed technique for making pre-flight and in-flight calibration checks on both piezoelectric and dc type accelerometers will be presented. A method for actually vibrating the piezoelectric accelerometer seismic mass over a wide frequency range and g level will be described in detail. The piezoelectric activity of the sensing element is obtained, and a meaningful calibration check may be observed immediately before count down flight. On the low-frequency accelerometers which incorporate magnetic damping, the damping cup and, hence, the seismic mass are displaced a known value of g with application of a dc voltage. This approach is used on potentiometer, variable reluctance, and differential transformer types of pickoffs.

N3. Performance Characteristics of a Wide-Range Mechanical Impedance Pickup. R. R. BOUCHE, *Endevco Corporation, Pasadena, California.*—In recent years there has been increased interest in measuring the mechanical impedance at a point on a structure. This paper describes the results of tests conducted during the calibration and evaluation of a mechanical impedance pickup of new design. The pickup weighs ½ lb and includes three force gauge elements and three accelerometers. The calibration results indicate the impedance pickup is suitable for use in the frequency range from ten to several thousand cycles per second. The effective end-mass of the pickup is less than two-tenths pounds and its stiffness greatly exceeds 10 million lb/in. These and other important characteristics indicate the pickup may be used in a wide range of applications requiring mechanical impedance measurements. The test results were obtained on sinusoidal calibration equipment and on a new type shock motion calibrator.

N4. Quartz Crystal Accelerometers and "Charge" Amplifiers. W. A. STUDIER (nonmember), *Kistler Instrument Corporation, North Tonawanda, New York.*—Measuring systems using these components have excellent sensitivity over wide ranges of temperature, permit precise measurements of accleration to 50 000 g through frquency range of 0.1 cps to 50 kc, and feature the capability of simple, direct calibration by quasi-static methods. Quartz crystal accelerometers are unsurpassed for linearity, complete freedom from hysteresis, negligible deflection under load, and ability to withstand severe overloads without damage. The recent development of the "charge" amplifier provides industry with a new and very useful tool. (Theory of operation, applications, and advantages are reviewed in detail.) Three calibration techniques are described. The extremely high insulation resistance of quartz transducers is exploited in the design of the "charge" amplifier to enable calibration at high working levels by direct application of weights to the seismic mass. After calibration in the "near-dc" mode, the system is returned to drift-free, dynamic operation with no loss of accuracy. Intermediate level calibration involves comparison with a laboratory standard servo-accelerometer; accelerations between plus and minus one g are calibrated by precise orientation of the instrument in the earth's gravity field.

N5. Zero Shift in Piezoelectric Transducers. HARTLEY J. JENSEN (nonmember), *Sandia Corporation, Livermore, California.*—"Zero Shift," as applied to acceleration measurements, is a term that refers to spurious changes in the recorded zero value. The changes are the result of faulty instrumentation response to the measured acceleration. Zero shifts caused by insufficient RC time constant, amplifier saturation, and cable effects, are well known and are not considered in detail. Zero shifts caused by spurious transducer output signals resulting from large magnitude shock (order of 1000 g) are of principal concern. Consideration is given to (a) separation of zero shifts caused by the transducer from those resulting from other causes; (b) the effect this phenomenon has on the accuracy of the indicated shock motion; (c) the various possible sources of the zero shift (progress in this area will include the work of various manufacturers as well as Sandia); and (d) a review of the experimental results obtained at Sandia using transducers designed for minimum zero shift.

N6. Techniques for the Rapid Estimation of Accelerometer Natural Frequencies. E. W. CLEMENTS (nonmember), AND M. STONE,* *U. S. Naval Research Laboratory, Washington 25, D. C.*—In order to determine the usable frequency range of an accelerometer, it is necessary that its natural frequency be known. Three simple methods for approximately determining natural frequency are described: measurement of the ringing frequency excited by electrical and mechanical pulse inputs to the accelerometer and measurement of the accelerometer's impedance as a function of frequency. While subsidiary to more refined calibration techniques, these methods are useful because of their speed and simplicity and because they may be utilized with the accelerometer in place. Results obtained for several piezoelectric accelerometers are given.

* At present with Louisiana State University.

N7. Effect of Nonaxial Motion on Accelerometer Calibration. E. R. SMITH (nonmember), W. S. EPSTEIN (nonmember), E. T. PIERCE, S. EDELMAN, AND E. JONES, *National Bureau of Standards, Washington 25, D. C.*—It is usual to monitor harmonic distortion, noise, and hum in the pickup signal during calibration but this is not enough to ensure suitable motion for accurate calibration. This paper describes calibration of an accelerometer on two shakers which were monitored for nonaxial motion. The nonaxial motion was detected by comparing the signals from three small pickups cemented equidistantly around the periphery of the shake table and the signal from the pickup being calibrated which was mounted at the center of the shake table. Each of the four pickups provided the signal for the vertical axis of a Lissajous figure. All of the Lissajous figures had a common horizontal signal which came from the oscillator driving the shaker through a calibrated phase shifter. A special eight channel electronic switch was used to put all of the Lissajous figures on the same CRO. During axial pistonlike motion all of the figures behaved similarly. During nonaxial motion the relative sizes and inclinations differed. The phase shifter provided a means for measuring differences in the phase of the motion at the pickup locations. Valid calibration points fell on a smooth curve. Points taken when the monitoring system showed nonaxial motion for a given shaker at a particular frequency fell off the curve while the calibration point taken at the same frequency with the other shaker which showed good motion fell on the curve.

Session O. Physical Acoustics

WILLIAM P. RANEY, *Chairman*

Contributed Papers

O1. Experimental Study of the Edge Tone Force and the Resulting Acoustic Field. HAPPY HUGH UNFRIED, *Aerosonics Laboratory, Department of Engineering, University of California, Los Angeles 24, California.*—The undulating stream in a jet edgetone system gives rise to a fluctuating pressure distribution over the surface of the wedge, and it is the integrated pressure over this surface which is called the edgetone force. In this series of experiments, the wedge dimensions have been chosen to be small compared to the corresponding acoustic wavelength so that time retardation effects over the surface of the wedge were minimized; hence, it is an effective point load which was measured. The wedge was supported on a pair of zirconate titanate transducers and this system was calibrated by passing a current through the stainless steel wedge placed in a magnetic field. The nozzle block and associated equipment were designed to minimize interference with the acoustic field. In the low-frequency region of operation, measurements show that the acoustic field of the edgetone can be represented by a point dipole whose magnitude corresponds to the measured force. The upper limit for this representation in terms of Reynolds and Strouhal numbers will be discussed. (Taken in part from a M.S. thesis, Dept. of Engr. UCLA, and in part from work supported by the Office of Naval Research.)

O2. Sound from the Action of Vorticity. ALAN POWELL, *Aerosonics Laboratory, Department of Engineering, University of California, Los Angeles 24, California.*—A formulation is suggested which shows how the sound from an aerodynamic flow can be primarily attributed to a volume distribution of dipole generators when a contiguous rigid body sustains a fluctuating fluid force. This can be interpreted in terms of the local fluid acceleration but, more significantly, can be attributed to the deformation of rings of vorticity in the flow (analogous to varying magnetic shells in electromagnetic theory). This leads to a new model especially applicable to aeolian- and edgetones. In the absence of a net fluctuating force, the sound from the vortex ring action degenerates from dipole to quadrupole nature. This situation is illustrated by considering the formation of a single vortex (of length l) from a length b of a thin layer of shear rate U, the distortion of the layer producing sound energy of the order of $U^7l^2ba^{-5}\times(2.10)^{-4}$. (This is also accountable for by a longitudinal quadrupole generation according to Lighthill's theory.) This must be supplemented by a comparable amount of energy resulting from the dilatational effect due to the appearance of the low pressure core of the vortex. The acoustic power emanating from a succession of such vortices is roughly $U^8l^2a^{-5}\times(2.10)^{-4}$. This point of view corresponds to the classical hydrodynamic result that the velocity (and hence pressure) at a point is derivable from a vector field (due to vorticity) together with a scalar field (associated with dilatation), except that time retardation effects must be recognized.

O3. Development of Edgetone Theory. ALAN POWELL, *Aerosonics Laboratory, Department of Engineering, University of California, Los Angeles 24, California.*—The tone produced when the jet of air issuing from a small slit strikes an edge has provided "a battleground for rival theories" during the past hundred years. This paper traces the development of understanding of the phenomenon in the light of recent developments. Some of the earlier ideas were along the right lines; for reasons now apparent, they failed to produce close agreement with experiment, with the result that other ideas sprang up (especially as a result of experimental observations), some of which were to fade away. However, the suggestions of a "preferential wavelength" for the stream disturbance and of feedback have survived to provide the foundations for present theories. The feedback mechanism has been described in terms of pressure fields and hydrodynamics (vortex) fields; the latter outlook is further developed here to show how these "rival" viewpoints are actually in close accord. Thus, a century has passed before the true threads can be recognized and synthesized; the edgetone now enters a new era of quantitative study.

O4. Low-Amplitude Nonlinear Theory of Plane Progressive Waves. David T. Blackstock, *Research Division, General Dynamics/Electronics, Rochester, New York.*—As the title suggests, this is a theory of plane waves of finite amplitude which is intermediate between traditional small-signal theory and exact nonlinear theory. It is based upon a certain assumption concerning the relation between wave amplitude and distance from the wave source. When this assumption holds, as it does in a large class of cases, one can simplify substantially Earnshaw's exact but highly unmanageable solution for plane waves. Application of the same assumption to the differential equations governing plane-wave propagation allows one to derive a useful approximate nonlinear wave equation. The exact solution of this approximate equation is the same as the simplified version of Earnshaw's solution. One interesting feature of this theory is that the form of the approximate nonlinear wave equation and its solution are the same in Eulerian and Lagrangian coordinates. A specific example will be discussed, namely the wave motion produced by sinusoidal motion of a piston in a lossless tube. It is found that the work of many past investigators can be unified by this theory. The limits of validity of the theory will also be discussed. (Part of this work was carried out at Harvard University with support from the Office of Naval Research.)

O5. Stokes-Navier Equations and the Fundamental Equations of Flow Noise. E. J. Skudrzyk and G. P. Haddle, *Ordnance Research Laboratory, The Pennsylvania State University, University Park, Pennsylvania.*—For ideal fluids, pressure may be considered to represent a function of the density that is defined by the state equation and the Euler equations. For viscous flow, variable pressure has to be replaced by the stress tensor. A scalar "pressure" no longer exists. The only scalar point functions that can still be retained for a viscous fluid are density and the trace of stress tensor (the average of the normal stresses in the three coordinated directions). The trace of the stress tensor and the density are, therefore, necessarily functions of one another. This conclusion makes it possible to expand (as Stokes has) the concept of pressure for viscous flow by defining it as the average value of the normal stresses. This generalized pressure is a function of the density and measures not the true stress in the fluid but the average of the stresses over all possible directions. In the classical theory of internal friction, the relation $\lambda+\frac{2}{3}u=0$ is wrongly impressed on the volume viscosity λ. In modern derivations, the volume viscosity λ is retained in its general form, but the pressure is no longer identified with a third of the exact value of the trace of the stress tensor. This situation makes it necessary to rederive the fundamental equations of flow and of flow noise; the derivations to be given are exact; the only assumptions that will be introduced are those of an isotropic fluid and of ideal fluid friction; the friction forces depend on the velocities only, and not on their higher-order time derivatives.

O6. "Incompressible" Acoustic Near Field as a Spatial Distribution of Sources Generating the Far Field. H. S. Ribner, *Institute of Aerophysics, University of Toronto, Toronto, Ontario, Canada.*—*Principle:* Suppose one calculates an "incompressible" flow (pressure field $p^{(0)}$) by substituting Laplace's equation for the wave equation in a given acoustic problem. Then the far field $p^{(1)}$ is a solution of the wave equation for a spatial distribution of matter sources of strength $-c_0^{-2}\partial p^{(0)}/\partial t$. (Boundary condition: zero normal velocity.) This is readily demonstrated. *Physical significance:* Since the fluid is actually compressible, isentropy gives $-c_0^{-2}\partial p^{(0)}/\partial t=-\partial \rho^{(0)}/\partial t$. This density fluctuation is precisely equivalent to a volume fluctuation of the fluid elements. *Pulsating sphere:* A pulsating sphere generating an "incompressible" field $p^{(0)}$ may be replaced by a rigid sphere surrounded by a cloud of sources of strength $-c_0^{-2}\partial p^{(0)}/\partial t$; this will generate the far field $p^{(1)}$. *Aeolian tones:* The principle helps explain nonvanishing generation of sound by a stationary rod in an air jet. The sound energy flow at a stationary surface must be zero: the sound is actually emitted from the region of fluctuating compression surrounding the rod. *Jet noise:* [J. Acoust. Soc. Am. **31**, 245 (1959)]. The "incompressible" approximation to the field within the jet is $p^{(0)}$. Source-like pulsations of the fluid elements in response to $p^{(0)}$ are considered to generate the far-field sound $p^{(1)}$. The source strength, modified for convection, is $-c_0^{-2}Dp^{(0)}/Dt$. (This work has been supported partially by the Air Force Office of Scientific Research of the U. S. Air Research and Development Command.)

O7. Reflection of Sound from a Nonuniform Periodic Boundary. H. S. Heaps, *Nova Scotia Technical College, Halifax, and Naval Research Establishment, Dartmouth, Nova Scotia, Canada.*—The concept of surface impedance is used to investigate theoretically the reflection of sound from a nonuniform boundary whose irregularities possess a periodic variation. The method used is similar to that of Lysanov [J. Acoustics USSR **4**, 47 (1958)]. By supposing the impedance to be a certain function of the angle of incidence, the results may be applied to determine the sound reflected from a sinusoidal interface between two propagating media. Numerical results are presented to indicate the dependence of the reflected radiation upon the form of the impedance variation, the angle of incidence, and the wavelength of the sound.

O8. Acoustic Radiation Impedance of a Rigid Rectangular Piston in a Moving Airstream. H. E. Plumblee, *Lockheed Aircraft Corporation, Marietta, Georgia.*—The effect of air flow on the radiation pattern of a simple source in a moving field, using the "retarded" and "advanced" potential wave equation solutions, is introduced into the classical derivation of radiation impedance. The impedance is derived for two regions of flow velocity: (1) subsonic, where only the "retarded" potential solution is required, and (2) supersonic, where both solutions are necessary. The impedance is derived in terms of the following normalized parameters: piston width/length ratio, (wave number)×(length), and Mach number. A useful application of the results is illustrated through the calculation of acoustic response of a rectangular cavity in a high-velocity airflow. From comparison with measured response, it is concluded that the radiation impedance solutions provide adequate definition of boundary conditions in the actual physical system. (This work has been sponsored by a Wright Air Development Division contract.)

O9. Acoustic Response of a Rectangular Cavity Excited by a Moving Airstream. L. W. Lassiter, H. E. Plumblee, and J. S. Gibson, *Lockheed Aircraft Corporation, Marietta, Georgia.*—A recess or cavity in an aerodynamic body in motion is usually subjected to intense acoustic and aerodynamic response. This response involves the combined effect of a random aerodynamic "buffet" and discrete acoustic resonances. A theory has been developed to define the acoustic resonant response, using the radiation resistance of a rectangular cavity as an intermediate step. The acoustic model of the cavity is that of a rectangular room with five walls rigid and one "soft." Comparisons of theoretical and experimental response are made at Mach numbers from 0.2 to 5.0. Movies are shown illustrating the boundary layer phenomenon associated with the resonant response of the cavity. (This work supported by a Wright Air Development Division contract.)

Session P. Speech Communications III

FRANKLIN S. COOPER, *Chairman*

Contributed Papers

P1. Investigation of the Precision of an Articulation Testing Program. CHARLES W. STUCKEY (nonmember), *Engineering Experiment Station, Georgia Institute of Technology, Atlanta, Georgia.*—A mathematical model of the individual articulation scores of a team of listeners during their training period was developed. A similar model was developed for a fully trained articulation team. By contrasting the two models, statistical hypotheses of whether or not a team had had sufficient training were formulated, and a method of testing these hypotheses was evolved. To test the validity of this method of evaluating training, an articulation team was formed and trained. It was concluded that the method of evaluation is valid and that after 20 hr the team was fully trained in terms of the criterion adopted. The effect of memorization on the articulation scores was evaluated and found to be negligible. The maximum variance of a trained team was found, and the precision of the results was computed as a function of the number of repetitions averaged. By averaging the scores from a sufficient number of repetitions, any desired degree of precision may be obtained. An articulation team is proved to be a very precise and reliable measuring instrument for ranking voice-communications equipment in terms of degree of acceptability.

P2. Concurrent Repetition of a Continuous Flow of Words. RUSSELL L. SERGEANT (nonmember), *U. S. Naval Medical Research Laboratory, Groton, Connecticut.*—This study was designed to evaluate the effect of rate (68, 127, 180 words/min) and intensity (20, 40, 60 SRT) of stimulus presentation, and of practice, upon concurrent repetition of a continuous flow of heard words. Subjects were 33 enlisted men. A three-part tape was made to obtain an SRT, to present a pretest trial, and to present three PB-50 lists read as continuous speech. Intelligibility scores were assigned to each subject's taped repetitions. Twelve subjects had an 8-day practice with other PB lists. Analysis of variance yielded an F associated with practice significant beyond the 0.5% level. The F's for rate and intensity were significant beyond the 0.1% level. Pearson r's were calculated between performance level and six psychological and auditory tests. Only the r of $+0.54$ with the Seashore Rhythm test was significant. It was concluded that: (a) Performance improves with increased intensity of heard speech, and with practice; (b) Performance increases when the rate of stimulus presentation is decreased; (c) A positive relationship exists between performance and scores on the Seashore Rhythm test; (d) Under proper conditions of rate and intensity, some persons can be trained to perform with more than 90% intelligibility.

P3. Analysis of Intelligibility of Tissue-Conducted Speech Signals. HERBERT J. OYER, *Michigan State University, East Lansing, Michigan.*—Previous research has revealed that acoustic energy resulting from vocal signals differs in sound pressure level as a function of (1) the anatomical sites at which signal pickup takes place, (2) the composition of the vocal signals, and (3) the fidelity of the recording systems. It has been shown experimentally also that vocal signals picked up out of the external auditory canals of speakers are significantly more intelligible at -15 db and -18 db S/N ratios than are signals simultaneously recorded at the lips. Phonetic analysis of the signals recorded at ear and lips points toward those sounds which cause the ear-recorded signals to be superior. They are the vowels *e*, ε, *i*. The present research is directed toward a determination of the relative intelligibility of words as they were recorded from six subjects representing two specific body types. Recordings were made at the following anatomical locations: (1) frontal bone of skull, 1 in. above nasalis, (2) mental protuberance of the mandible, (3) immediately below superior thyroid notch of the laryngeal prominence, and (4) at the angle of the mouth. Results of intelligibility testing point toward the use of anatomical sites in addition to the lips for recording of voiced signals. (This research was sponsored by an All-University research grant No. 2771, Michigan State University.)

P4. Study of Intelligibility Improvement with Out-of-Phase Connections of Headphones. D. GREENBERG (nonmember),* AND H. O. BENECKE (nonmember),† *U. S. Naval Air Development Center, Johnsville, Pennsylvania.*—Previous work on the effect of interaural phase relations on the masking of speech by white noise, suggested the possibility of improving communication in ambient noise by simply reversing the connections of one earphone of the headset to produce out-of-phase reception. A series of intelligibility tests was conducted at the U. S. Naval Air Development Center, Johnsville, Pennsylvania, in 1949 to determine whether improvement could be obtained under conditions simulating those encountered in naval aircraft. Headphones in use then were almost entirely connected in-phase. The results of intelligibility tests, performed with listeners subjected to 119 db of simulated reciprocating-engine aircraft noise, showed that for low and intermediate signal levels, the out-of-phase scores averaged only about five percentage points higher than in-phase scores; for high signal levels, the out-of-phase scores were about five percentage points less. The overall results did not indicate any significant advantage for the out-of-phase connection for communications in naval aircraft.

* Now with Radio Corporation of America, Camden, New Jersey.
† Now with Radio Corporation of America, Moorestown, New Jersey.

P5. Measurement of Speaker Recognition. L. V. SURGENT (nonmember), AND L. H. YOST (nonmember), *General Dynamics/Electronics, Rochester, New York.*—A procedure was developed for assessing the relative capabilities of voice transmission systems with respect to "speaker recognition." Sets of five speakers each are composed randomly from a larger population. A listener is trained by an anticipation method to identify, by fictitious names, the members of a set of speakers, as heard over one of the systems to be evaluated. Following training, five additional speakers ("strangers") are added to the initial five and all ten presented in random order over the same or a different system. Listeners are tested to see if they can (a) distinguish the five "strangers" from the five members of the initial set (IFF-type speaker recognition) and (b) name the five speakers met during training (speaker identification). A different sentence (approximately 10-sec duration) is used for each presentation of a set of speakers in both training and testing. Experimental designs for comparing two or more systems simultaneously will be discussed. The method will be compared with other approaches. (The study on which this paper is based was sponsored by ARDC and RADC under contract.)

P6. Automatic Talker Recognition Using Time-Frequency Pattern Matching. SANDRA PRUZANSKY AND PETER D. BRICKER, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—A study of machine recognition of talkers has been carried out by digital computer simulation. Test materials consisted of several utterances of the digits one through ten spoken by five male talkers. The talkers were instructed to pause after each utterance, and an appropriate threshold for locating the beginning of each utterance was determined by studying the total energy displayed for each time section. These utterances were converted to time-frequency patterns of spectral energy. The dimensions of these patterns were 17 frequency bands by 60 time sections (1/70 sec each), and the entry at each array point was one of the 1024 possible energy levels. Three-utterance reference patterns were formed by adding together corresponding array points of each of three utterances spoken by the same talker, making five reference patterns for each digit. Talker recognition was accomplished by cross correlating the array of the remaining single utterances of each digit with each of the five reference arrays for that digit and selecting the talker with the maximum correlation. Typical results, for patterns with full bandwidth, indicated a small percentage of errors in talker recognition. Talkers were correctly identified for 94 out of 100 test utterances (two utterances of each digit by each speaker). The errors were distributed throughout the ten digits. Variations in results for patterns which have been converted to a standard duration by a change of time scale and patterns with reduced bandwidth will be discussed.

P7. Decision Functions for Voiced-Unvoiced-Silence Detection. C. C. LOCHBAUM, E. E. DAVID, JR., AND M. V. MATHEWS, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—At the June, 1960, Acoustical Society of America meeting, we described a voiced-unvoiced-silence detection procedure. It incorporated two independent threshold decision rules, one operating on the zero moment of the speech spectrum for the spoken-silent choice, the other operating on the first moment for the voiced-unvoiced choice. A generalization of this procedure involves a decision which depends jointly on the two moments. This paper compares the performance of this process with the simpler one. Value functions placing specified weights on the nine possible outcomes of this 3×3 detection situation were defined, and decision rules maximizing the values were computed. Maximum likelihood, maximum likelihood weighted by prior probabilities, and rules tailored to vocoder applications were studied. Probabilities of the six possible confusions and the three correct answers were computed for all rules. Data for the evaluation consisted of twenty 3-sec utterances by two men and two women. Voiced, unvoiced, and silent intervals in these speech samples were established visually from spectrograms. Performance depends, of course, on the value functions, but at most a reduction of ⅓ in the over-all error rate over the simple rule was achieved.

P8. Speech Sound Classification Based on Signal Statistics. R. BAKIS (nonmember), *International Business Machines Corporation Research Center, Yorktown Heights, New York.*—A method of quantizing the output of a vocoder-type speech analyzer is described. This quantization effectively classifies the sound at each instant into one of a small number (typically, less than 100) of classes, but in such a way as to allow relatively accurate reconstruction of the original unquantized voltages The boundaries of the classes are chosen on the basis of the statistical distribution of the voltages, using redundancies which may exist to reduce the number of classes to a low value. Experimental results are presented, illustrating classification schemes resulting from this procedure. These schemes bear some resemblance to conventional phonetic analyses of speech sounds, but they have the advantage of being derivable by purely mechanical means, without the use of human intuition or judgment. The use of such sound classification for automatic speech recognition is discussed.

Session Q. Audio Engineering and Electroacoustics

WALTER KOIDAN, *Chairman*

Contributed Papers

Q1. Phasor Analysis of the Stereophonic Phenomena. BENJAMIN B. BAUER, *CBS Laboratories, Stamford, Connecticut.*—An improved understanding of the stereophonic phenomena may be obtained by use of acoustical pressure phasors to portray the sound pressures at the ears of the observer. Pressure phasors explain the existence and the nature of the virtual image formation by two loudspeakers, the effect of out-of-phase signals, the effect of observer head motions, the effect of being off the center of symmetry, and the action of a circuit for converting binaural earphones to stereophonic listening.

Q2. Further Progress with Colorless Artificial Reverberation. M. R. SCHROEDER, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—In a previous paper [J. Acoust. Soc. Am. **32**, 1520 (1960)], artificial reverberators having flat amplitude-frequency responses were described. Such "flat" reverberators preserve the amplitude spectrum of the reverberated sound, thus avoiding any undesirable "coloration." In the present paper, an alternative approach to colorless artificial reverberation is described in which several "combfilters" (delay lines with multiple reflections) connected in *parallel* are employed to create a *subjectively* colorless reverberation. Such reverberators do not have a flat frequency response in the physical sense, but are nevertheless subjectively indistinguishable from flat reverberators if the following (sufficient) conditions are met: (1) The number of combfilters must be at least three. (2) Their delays must be substantially different and incommensurate. (3) The average distance $\langle\Delta f\rangle$ between adjacent relative maxima of the frequency response of the reverberator must be smaller than about 5 cps. (This condition is equivalent to requiring that the sum total of the round-trip delays employed in the combfilters exceed 200 msec.) (4) For reverberation times T in excess of 2 sec, the product $\langle\Delta f\rangle\cdot T$ must be smaller than 10. To meet the additional requirement of a flutter-free transient response, the reverberator must furnish at least one echo per msec for times in excess of about 100 msec after the direct sound. Finally, for the realistic simulation of large halls, it is recommended that a delay of the order of 40 msec be introduced between the direct sound and the reverberation. The advantage of the present "combfilter approach" to artificial reverberation is that it allows greater flexibility in mixing direct and reverberated sound. Also, the delay lines for the combfilters do not have to meet specifications as stringent as those employed in flat reverberators. A tape recording of artificially reverberated organ music will be played.

Q3. Design of Distributed-Loudspeaker Speech Systems. Martin F. Gardiner and David H. Kaye (nonmember), *Bolt Beranek and Newman Inc., Cambridge, Massachusetts.*—This paper considers the relation of various design parameters of distributed-loudspeaker speech-amplification systems to the two principal design criteria: "intelligibility" and "naturalness." A convenient design procedure is described involving the calculation of the speech "articulation index" for the system. Some of the results of recent experiments on the evaluation of distributed-loudspeaker speech systems will be included.

Q4. Improved Instrumentation for Impulse Noise Analysis. George J. Harbold (nonmember), and James W. Greene, *U. S. Naval School of Aviation Medicine, U. S. Naval Aviation Medical Center, Pensacola, Florida.*—A longitudinal study of impulse-type noise is being conducted by the Department of Neurophysiology and Acoustics, U. S. Naval School of Aviation Medicine, Pensacola, Florida. As a part of the study, efforts were directed toward improvements in the techniques and instrumentation usually used to quantify the rise times, peak pressure levels, and decay patterns of impulse noises. This paper reports an improved instrumentation for impulse noise analysis. The limitations imposed by condenser microphones, magnetic recorders, and level recorders in response to transients associated with blast phenomena were investigated. In view of these serious limitations, the use of alternative techniques and systems was explored. The use of high-intensity crystal microphones and a storage-type scope was found to provide sufficient accuracy and flexibility for the variety of explosive transients encountered in field studies of many weapons. Pictorial evidence will be presented to illustrate the extent to which the limitations of earlier systems were overcome.

Q5. Measurement of Probability Density. Herbert L. Fox, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The increasing requirements for probability density measurements of the amplitude characteristics of time-varying signals require a careful examination of (1) requirements for probability density analyzers, (2) techniques of measurement, and (3) uncertainty in estimates obtained from samples of finite duration. Requirements discussed include bandwidth and dynamic range, amplitude sampling interval (window width), and necessary readouts. Various techniques of using a probability density analyzer are discussed, with emphasis on the necessity of using different techniques for different types of data and/or use thereof. A method of scanning that guarantees constant uncertainty over the entire range of the probability density is described. An expression for the uncertainty of the measured probability density as a function of sampling time is obtained, and measurements are shown indicating the relationship of sampling time, window width, bandwidth, and probability density to the uncertainty. The normalized variance is shown to be of the form $\sigma^2=K/[f_0T\Delta x w(x)]$. The constant K is derived, using alternate assumptions, and is evaluated experimentally. The analysis in this paper is supported by measurements made with, and on, a commercial probability density analyzer.

Q6. An Automatic One-Third Octave Band Analyzer for Noise and Vibration Analysis. G. F. Ginty (nonmember), and R. D. Collier, *General Dynamics Corporation, Electric Boat Division, Groton, Connecticut.*—The problem of rapidly analyzing large volumes of acoustic data in analog form has been successfully met with a semiautomatic system. The principal feature of the system is the ability to process signals simultaneously in 31 parallel $\frac{1}{3}$-octave filters over a frequency range of 10 to 10 000 cps over selected time periods. This is accomplished through storage capacitors, and provides a choice of two averaging times for average-amplitude modulated signals or peak value. A rotary selector stepping switch scans capacitor voltages sequentially, applying them to a dc logarithmic amplifier. The output is plotted on a Mosley X-Y plotter in the form of a bar graph. The instrument occupies a standard 6-ft rack. Sample results from vibration and sound-pressure level analyses are discussed.

Q7. Extension of the Pressure Response-Frequency Characteristics of Condenser Microphones by Means of Carrier-Frequency Techniques. Walter Koidan and June O. Magruder (nonmember), *Sound Section, National Bureau of Standards, Washington 25, D. C.*—A method was investigated for extending the frequency range of microphone pressure calibrations to infrasonic and ultrasonic frequencies when the response at a single frequency is known. Assuming the microphone to be reciprocal, its pressure response is $(u/i)_{p=0}$, where u is the diaphragm volume velocity, i the driving current through the electrical terminals, and p the sound pressure directly in front of the diaphragm. The diaphragm of a condenser microphone is driven by impressing a voltage across its electrical terminals, and its volume displacement is observed simultaneously by means of a carrier-frequency circuit. A substitution technique is then applied using a variable capacitance diode, which makes it possible to determine accurately the shape of the pressure-response curve as a function of frequency. No microphone other than the one to be calibrated is required. For frequencies at which the acoustic impedance of the microphone is not very much higher than its radiation impedance, the required acoustic termination is provided by a quarter wavelength tube. The experimental results are compared with coupler reciprocity calibrations performed from 1 cps to 20 000 cps. Limitations of the electrostatic actuator also are discussed, and a calibration method is described which uses a quarter wavelength tube in conjunction with an actuator to reduce the acoustic impedance seen by the diaphragm.

Q8. A Wide-Band Carrier System for Condenser Microphone. Herbert L. Fox, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—A carrier microphone system has been developed for measuring very low sound pressure levels over a bandwidth from static pressure to 100 000 cps. The system is suitable for use with conventional condenser microphones and other types of capacitive transducers. This paper discusses (1) the problems of carrier systems vs direct bias systems in achieving low noise in the low- and high-frequency region, (2) a method of using tuned circuits at the carrier frequency without restricting the input bandwidth to the Q of the individual tuned circuits, and (3) the use of electrostatic methods of actuation to obtain equivalent pressure calibration in a carrier system. An equivalent acoustic noise floor less than Fletcher's minimum audible field curve over the entire audio bandwidth has been achieved. Pressure calibration of condenser microphones to 100 kc has been accomplished with the system described. (This work was supported in part by a U. S. Air Force contract.)

Q9. A Miniature Universal Lightweight Military Microphone. E. D. Simshauser and R. M. Carrell, *Defense Electronics Products, Radio Corporation of America Inc., Camden, New Jersey.*—A group of moving-coil noise-cancelling microphones has been developed which exhibits a sensitivity approximately 6 db greater than previous microphones of equivalent size, weight, and bandwidth. This gain is the result of an asymmetrical internal construction which results in cardioid directional characteristics. A result of this construction is a microphone which is small enough for an oxygen helmet and light enough for boom use, having a sensitivity of 40 μv/d cm² with a 150-omb load, which is compatible with Army and Navy microphone circuits. The equivalent circuits and design philosophy are described. (The work was supported by a U. S. Air Force contract.)

Q10. Low-Frequency Microphone. J. R. Barger (nonmember), A. Witchey, and L. Weinreb, *Defense Electronic Products, Radio Corporation of America Inc., Camden, New Jersey.*—This paper describes the design of a high-sensitivity low-frequency microphone. It has two basic components, a microphone element and a preamplifier. The microphone element is of the condenser type, using an edge-clamped stretched-Mylar diaphragm. The condenser microphone is used as one arm of a capacitance bridge which is furnished excitation by an rf oscillator. The bridge output is an rf signal modulated by the variations in microphone capacitance caused by pressure changes in the incoming signals. After suitable amplification, the modulated signal is processed in a peak detector. The resulting low-frequency signals are directly coupled into an emitter follower output stage. This pressure-sensitive transducer provides a useful output at pressure levels from 0.01 to 40 d/cm² and frequencies of 0.1 to 100 cps. The frequency response of the microphone can be varied to suit system requirements by minor adjustments of either the microphone element or the preamplifier. Differences of phase shift between two or more microphones can be held to less than 1° at 20 cps. (The work reported herein was supported in part by a U. S. Army Signal Corps contract.)

Q11. Lagrangian Formulation and Network Analogs for Multi-Mode Electromechanical Transducers. Frank J. Rosato, *Applied Research Laboratory, Sylvania Electronic Systems, Waltham 54, Massachusetts.*—The Lagrangian equations which are useful in the solution and interpretation of the electrodynamic characteristics of multimode transducers of the piezoelectric and magnetostrictive types are presented. In order to bring such electromechanical systems within the framework of the Lagrangian theory, use is made of thermodynamic potentials defined for piezoelectricity and magnetostriction. [A table of such potentials for the piezoelectric case is given in the book by W. P. Mason, *Piezoelectric Crystals and Their Application to Ultrasonics* (D. Van Nostrand Company, Inc., Princeton, New Jersey, 1950), p. 34.] The two most useful potentials for Lagrangian application are found to be the internal energy density U and the electric enthalpy H_2. These potential functions, together with the kinetic energy function, are used in constructing Lagrange's equations for electromechanical systems. These equations are essentially equations of dynamic equilibrium constructed in terms of the kinetic and potential functions, which are expressed in terms of generalized coordinates. They are shown to lead to a system of electromechanical equations for the transducer which have a structural resemblance with the mesh and nodal equations of coupled electric networks. Hence, they may be represented by electric network analogs wherein the mechanical terminal variables are generalized force and generalized velocity. These network analogs are shown to offer several distinct advantages over the conventional filter-type analog networks: They are insensitive to the location or distribution of the external forces, they exhibit explicitly the characteristic modes and the electromechanical coupling to each mode, and they lend themselves more readily to analog electrical measurements. The analog networks obtained for several transducers will be presented. (Work performed by the author at the Acoustics Research Laboratory, Harvard University, sponsored in part by the Office of Naval Research.)

Session R. Psychological Acoustics II

J. C. R. Licklider, *Chairman*

Contributed Papers

R1. Detecting a Signal in Noise: Further Considerations. Lloyd A. Jeffress, *Department of Psychology and Defense Research Laboratory, The University of Texas, Austin 12, Texas.*—In discussing the masking of tonal signals, L. A. Jeffress, H. C. Blodgett, T. T. Sandel, and C. L. Wood [J. Acoust. Soc. Am. **28**, 416 (1956)] likened the auditory mechanism for detection to the kind of equipment a physicist might use in attempting to perform the same task. Both involve the use of a narrow filter followed by a detector. The earlier paper described the detector as needing "to be sensitive only to changes in over-all level and to have a certain time characteristic." The present paper is concerned with properties of the detector, especially its time characteristics for both monaural and binaural stimulation. Some of the problems of signal detection will be illustrated by oscillographic pictures of narrow-band noise and signals for both the homophasic and the antiphasic conditions. (This work was supported by the Bureau of Ships.)

R2. Method of Free Response in Signal Detection: A Simplified Technique. James P. Egan, *Hearing and Communication Laboratory, Indiana University, Bloomington, Indiana.*—The "method of free response" refers to the following procedure. In a background of continuous noise, a weak signal is presented several times in a long (2-min) observation interval. The temporal intervals between the occurrence of the signals are randomly selected. No further indication is given to the listener as to the time at which a signal occurs. Over a series of observation intervals, the listener adopts various criteria, and he presses a single "yes-key" whenever he thinks he hears a signal. In our previous studies, the relation between rate of response and time after signal (with the criterion as the parameter) was determined in fine detail. Such an analysis is so tedious (or expensive) that the following simplified technique is suggested. The 2-sec interval just following a signal is compared with a similar interval that occurs considerably after a signal. The proportion of such intervals containing a response is determined for both the near and far intervals, and these proportions are used instead of D (detections) and O (operant level). Only three counter readings are required. The relation between D and O is secured by inducing changes in the criterion of the listener. Estimates of the error introduced by this simplified procedure are presented. (This research was supported by the Operational Applications Office, Air Force Cambridge Research Center.)

R3. Simultaneous vs Successive Observation Intervals in Signal Detection. S. M. PFAFFLIN (nonmember), *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Performance in a two-interval forced-choice signal detection experiment employing simultaneous presentation of the observation intervals was compared with the performance obtained when the observation intervals occurred successively in time. In the simultaneous condition, two independent samples of random noise, one of which contained a sinusoidal signal, were introduced into the observer's earphones at the same time, one sample to each ear. The observer's task was to indicate into which of his ears the signal had been introduced. In the successive observation condition, the bursts of noise were separated by a short time interval and the observer was to indicate whether the first or the second burst contained the signal. Detection under the two methods of presentation varied for different subjects and signal parameters, but, in general, performance during simultaneous presentation was inferior to that obtained during successive presentation. Although a gain in detectability might have been expected to result from the opportunity for direct stimulus comparison provided by the simultaneous method, these data show that any such gain is more than offset by the introduction of factors detrimental to detection.

R4. Binaural Detection of Single Frequency Signals in the Presence of Noise. MARK B. GARDNER, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—It is well known that when signal and noise sources occupy different spacial positions, detection of the signal is influenced by the orientation of the observer. This paper presents a study of these and related factors as they affect the detectability of single frequency signals using a binural transmission system. An adjustable-rate pulse-tone method of signal presentation was employed. The two microphones were supported in free space both with and without an artificial head. For the former, various "heads" were used to provide "interaural" separations of 2.5–13 in., while for the latter, where phase differences alone are present, data were taken for linear and rotational separations of the microphones. The results indicate that over-all detectability is a function of the combined binaural phase relationships of both signal and noise and of the monaural intensity relationship of the two signals at the ear having the higher signal-to-noise ratio. In agreement with other studies, the importance of phase differences was found limited to frequencies below about 2000 cps while the importance of pressure-level differences increased as a function of frequency above about 250 cps.

R5. Head-Position Identification. H. N. WRIGHT, *C. W. Schilling Auditory Research Center, Inc., Groton, Connecticut.*—A binaural recording of traffic sounds that reached an artificial head oriented in five different positions was presented to five subjects, each of whom responded under four different criteria. A Type II ROC curve for the five listeners was linear, $P(Y|C)=0.60\ P(Y|I)+0.44$, and therefore consistent with a guessing model and inconsistent with the detection theory model. Further analysis of the results showed that performance (percent correct and information transmitted) was independent of criterion and did not change systematically over successive trials. (This work was supported by contract from the Office of Vocational Rehabilitation.)

R6. Learning to Identify Nonverbal Sounds. JOHN A. SWETS, DAVID M. GREEN, AND EARL F. WINTER, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Research is under way to determine whether or not the techniques of automated teaching (active participation of the learner, immediate feedback, presentation of items conditional upon the learner's recent past performance, etc.) will significantly facilitate learning to identify nonverbal sounds. We suspect that parametric study of the lesson variables will define a teaching procedure that produces results surpassing the limitations on identification ability disclosed by previous research. This paper describes the research design and apparatus, and the results of the experiments conducted to date. In this study, a PDP-1 digital computer is used to realize a large number of teaching machines—the program permits the experimenter to set up a lesson in any format desired. The computer exercises simultaneous control over independent lessons for several subjects, and produces a complete record of the lesson as well as analyses of the data at the completion of the lesson. The computer also generates the sounds to be learned; it constructs any of a very large number of sounds with an average access time of 0.2 sec. Variables under study include the procedure for selecting the sound to be presented on a given trial, the reinforcement probability, the kind of feedback, the nature of the sounds, and the mode of response. (The research reported here is supported by the U. S. Navy Training Devices Center, Port Washington, New York, under contract for a Human Engineering project.)

R7. Signal Presentation Rate and Auditory Vigilance. RICHARD L. MARTZ AND J. DONALD HARRIS, *U. S. Naval Medical Research Laboratory, Groton, Connecticut.*—The probability of detection of visual signals by human observers in the vigilance situation has been shown to be a function of the rate at which signals are presented, where signals are given a high initial probability of detection along some (suprathreshold, intensity) gradient. This effect of signal presentation rate has not been directly demonstrated with auditory signals at threshold intensity. In a traditional vigilance design, six Navy men and six housewives gave aperiodic threshold determinations by Elliott's ascending method of limits for a 1000 cps tone in continuous broadband noise of 70 SPL during 48-min sessions. Following a 12-item pretest at the signal rate of 120 signals/hr, given at the beginning of each session to determine "relative zero," subjects were individually tested on one of six signal rates (2.5, 7.5, 15, 30, 60, and 120 signals/hr) daily for six days, with coefficients of variation [(variance)$^{\frac{1}{2}}$/mean] for intersignal intervals maintained at 1/15. Each determination involved n 2-db steps, where interstep intervals were approximately 5 sec, and no criterion or correction was set for false positives. Analyses of variance of mean relative intensity, latency, and distribution of false positive responses showed no consistent trends.

R8. Information in Simple Multidimensional Speech Messages. JOHN C. WEBSTER, *U. S. Navy Electronics Laboratory, San Diego 52, California.*—Simultaneous pairs of four-bit messages were presented to listeners at rates of two pairs every four seconds (4 bits/sec), three pairs every four seconds (6 bits/sec), four pairs every four seconds (8 bits/sec), and six pairs every four seconds (12 bits/sec). Each message of the pair was either [*a*] or [*i*], said by a male or a female, as a question (rising inflection) or a statement (falling inflection), and heard in either the right or the left ear. Sequences were made that were balanced (1) between ears, (2) between male and female voices, (3) between [*a*]'s and [*i*]'s, and (4) between questions and statements. The subject sometimes listened to both messages but usually listened only for the messages (1) in one ear, (2) of one voice, (3) of one vowel sound, or (4) of one inflection pattern. In each case he categorized all four dimensions. The best subject could receive correctly about six bits per second whether this was one of two messages at the 12 bit/sec rate or both of two at the 6 bit/sec rate. It was easiest to separate messages by ear, followed by voice (male or female), and vowel. It was difficult to separate or classify messages by inflection pattern. (Carried out at Applied Psychology Research Unit, Cambridge, England, on a National Science Foundation fellowship.)

Session S. Ultrasonic Physics

THEODORE A. LITOVITZ, *Chairman*

Contributed Papers

S1. Determination of Elastic Constants of Isotropic Materials at Megacycle Frequencies. THRYGVE R. MEEKER (nonmember), AND ALLEN H. MEITZLER, *Bell Telephone Laboratories, Inc., Whippany, New Jersey.*—The frequency dependence of the phase and group velocities for longitudinal and flexural waves in a plate provides the basis for several convenient methods of determining elastic constants. In these methods two independent elastic constants are determined in strip-shaped samples from measured values of certain special frequencies and delay times. One set of data is determined by the frequencies at which certain pairs of modes have either equal group or phase velocities. At other frequencies the delay vs frequency characteristics of various modes have extrema, and the frequencies at which these occur along with the measured delays provide another set of data. Theoretical calculations have been carried out which show the dependence of these measured quantities on Poisson's ratio. The shear wave velocity and Poisson's ratio are determined for a sample behaving essentially as an isotropic, elastic solid by selecting values that bring calculated and measured quantities into agreement. The methods to be discussed are capable of detecting anisotropic or anelastic behavior if either or both of these complications are present and have been used successfully at frequencies up to 35 mc.

S2. Attenuation of High Frequency Elastic Modes of Propagation in Strips of Polycrystalline Metals. ALLEN H. MEITZLER, *Bell Telephone Laboratories, Whippany, New Jersey.*—Longitudinal and flexural waves in an infinite plate of an isotropic elastic material can be analytically synthesized from a single dilatational potential function and a single rotational potential function. If A is the amplitude of the dilatational potential function and B is the amplitude of the rotational potential function, it is possible to calculate the ratio A/B for a particular mode as a function of frequency once the values of the roots of the appropriate frequency equation are at hand. Calculations of this kind have been carried out for the lowest three longitudinal and lowest three flexural modes and show that, in general, these modes are composite wave motions involving varying combinations of dilatational and rotational components such that A/B varies in amplitude and phase in a given mode as a function of frequency. The primary purpose of this paper is to introduce the "character" of the various modes, defined as $[(A/B)(A/B)^*]$, as a concept useful in understanding the selective attenuation of high frequency elastic modes in polycrystalline metal strips. In this type of medium, rotational waves are attenuated appreciably more than dilatational waves; hence a given mode might be expected to exist more strongly over the frequency range in which the total wave motion is predominantly dilatational in character. Experimental data for the attenuation of elastic pulses having 5 to 35 mc carrier frequencies and traveling in strips of polycrystalline aluminum, 5 to 10 mils thick, will be presented. In the results, a good correlation is observed between the frequency range over which a given mode is observed and the frequency range over which, theoretically, the character of the mode is large; i.e., $[(A/B)(A/B)^*]>1$.

S3. On Elastic Constants and Ultrasonic Velocities in Some Crystal Groups. WALTER G. MAYER AND PAUL M. PARKER (nonmember), *Physics Department, Michigan State University, East Lansing, Michigan.*—Expressions are given relating ultrasonic velocities in single crystals to the elastic constants. Without introducing approximations these expressions can be used in a relatively simple manner to evaluate the elastic constants. As an example the procedure is outlined for the trigonal system, based on existing velocity measurements in corundum. Some of the general expressions indicate certain invariance properties of the velocities in simple crystal groups. These properties can be used to check the results of certain velocity measurements or to determine the velocity of certain modes of vibration in some crystallographic directions.

S4. Ultrasonic Measurement of the Photoelastic Constants of ADP. K. ACHYUTHAN (nonmember), *Department of Physics, Michigan State University, East Lansing, Michigan.*—An ultrasonic technique, first suggested by Mueller [Z. Krist. **A99**, 122 (1938)], has been used to determine the ratios P_{12}/P_{11}, P_{13}/P_{33}, P_{31}/P_{11} of the photoelastic strain constants of ADP crystals. These values, along with the values of some of the stress (Carpenter, Ph.D. thesis, Harvard University, 1951) and elastic constants, have been used to determine a complete set of photoelastic constants. Accuracy of the measurements will be discussed. (This work was supported by the National Science Foundation.)

S5. Third Order Elastic Moduli of Germanium. W. P. MASON, H. J. MCSKIMIN, AND T. B. BATEMAN (nonmember), *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Measurements have been made for all 6 third-order elastic moduli of germanium by measuring ultrasonic velocities in selected directions when directed static stresses are applied to the crystal. Three measurements are obtained by using hydrostatic pressures, three by using a static compression along the ⟨001⟩ axis, and six by stressing the ⟨110⟩ axis with measurements being made along with the ⟨001⟩ direction and the ⟨1$\bar{1}$0⟩ direction. By using the finite strain formulas of Murnaghan, the measured velocities are related to the 3 second-order elastic moduli and the 6 third-order elastic moduli for a cubic crystal. The 12 sets of measurements provide considerable overlap and the probable errors are shown to be moderate. The values obtained are $C_{111}=-1.22\pm0.08\times10^{12}$ d/cm^2; $C_{112}=-1.450\pm0.15$; $C_{123}=+2.16\pm0.21$; $C_{144}=-0.17\pm0.19$; $C_{166}=-6.07\pm0.28$; $C_{456}=-1.65\pm0.19$.

S6. Some Measurements of Wave Velocities and Elastic Moduli for Cadmium Sulphide. H. J. MCSKIMIN, T. B. BATEMAN (nonmember), AND A. R. HUTSON (nonmember), *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—Because the literature indicates a rather large spread of values for the elastic moduli of single crystal cadmium sulphide, additional measurements made on a specimen of low conductivity and on one of high conductivity are reported in this paper. The effect of light on certain of the measured wave velocities in the former will be discussed briefly. The method of estimating uncertainties is detailed, and a procedure for removing the ambiguity in the sign of c_{13} is discussed. In units of 10^{11} d/cm² and for a temperature of 25°C the zero field elastic moduli are: $c^E_{11}=8.581$, $c^E_{12}=5.334$, $c^E_{13}=4.615$, $c^E_{33}=9.370$, $c^E_{44}=1.487$. Estimated uncertainties are less than 0.2%.

S7. Method for the Pressure Calibration of Pulse Operated Ultrasonic Transducers. WILLIAM W. LESTER, *Department of Physics, Michigan State University, East Lansing, Michigan.*—An optical diffraction method for the pressure calibration of pulsed ultrasonic transducers in transparent media is discussed. Results are presented which compare this method with optical calibration by means of continuous waves. As an example of the usefulness of the method, a 5-Mc ceramic transducer of the barium titanate type has been calibrated in water over the temperature range 5–33°C, using 3- to 10-μsec, 5-Mc input signals. Because of the low average input power, stable transducer temperatures are maintained, with no difficulty from streaming or heating of the liquid. (This work was supported by the Office of Naval Research.)

S8. Experimental Studies of the "Least Stable Waveworm." M. A. BREAZEALE, *Department of Physics, Michigan State University, East Lansing, Michigan.*—The "least stable waveform" of a finite amplitude wave is that in which the maximum pressure gradient is in the trailing edge of the wave. It can be formed by reflecting a distorted finite amplitude wave from a resilient boundary. By using a pulse technique, the increase of distortion of a finite amplitude wave, and then on reflection from a resilient boundary, the decrease of distortion, are shown. The effects of resilient and rigid boundaries on the diffraction of light by standing ultrasonic waves are demonstrated. For comparison, diffraction patterns for finite amplitude progressive waves are shown. (This work was supported by the Office of Naval Research.)

S9. Mutual Interactions of High Energy Ultrasonic Beams. THOMAS D. SACHS, *Ultrasonic Laboratory, Western Reserve University, Cleveland 6, Ohio.*—The conflicting claims as to production of nonlinear interactions between sound beams arising from second order terms, as presented by Ingard [J. Acoust. Soc. Am. 28, 367 (1956)] and Westervelt [J. Acoust. Soc. Am. 29, 199, 934 (1957)], were investigated by a pulse method. Time separation of initial beams and scattered field was possible using appropriate geometry. The scattered field was investigated for 14 different relative beam angles including 0 deg. The incident 1010 kc and 1410 kc beams had space-average peak pressures of 4.2 and 6.9 atm, respectively, with a minimum interaction volume of 150 cm³. Sensitivity was sufficient to detect interactions at least 70 db below those predicted by Ingard. No interactions whatever were found in spite of the divergent character of the initial beams. (This research was done in the Physics Department of the University of Innsbruck, Austria, in partial fulfillment of the requirements for the Ph.D. degree.)

S10. On the Derivation of Molecular Information from Ultrasonic Relaxation. J. E. PIERCY, *Division of Applied Physics, National Research Council, Ottawa 2, Ontario, Canada.*—A review of energy differences ΔE between rotational isomers of eight ethane derivatives reveals almost no correlation between recent apparently reliable ultrasonic values and those available from other techniques. The properties of the eight molecules have therefore been examined to ascertain how closely they conform to the hopeful assumptions inherent in the usual methods of deriving ΔE from the ultrasonic measurements. The normal neglect of the difference in dipole moment and volume between isomers and assumptions concerning undetermined static parameters are found to introduce errors in the derived values of ΔE which are at least the same magnitude as its absolute value. Similar errors may also be expected for other isomeric systems, and some are probably unavoidable for measurement in pure liquids, but can be avoided in favorable cases by solution measurement. Perhaps the most interesting discovery is the error which results from deviations from ideal mixing caused by a difference in dipole moment between isomers and the use of the Herzfeld-Kneser formula for thermal relaxation, which assumes ideal mixing. Much larger errors may be expected to result from nonideal mixing in the use of the Hall formula for structural relaxation in associated liquids.

S11. Ultrasonic Temperature Determinations in a Plasma. E. H. CARNEVALE, H. POSS (nonmember), AND J. M. YOS (nonmember), *Avco Corporation, Research and Advanced Development Division, Wilmington, Massachusetts.*—An ultrasonic pulse technique, similar in principle to that developed by Livengood, Rona, and Baruch [J. Acoust. Soc. Am. 26, 824 (1954)] for measuring gas temperatures in an internal combustion engine, has been used in determining the temperature of an electric arc plasma jet. Measurements of sound velocity at 1 mc/sec were obtained in high temperature air, argon, and helium at atmospheric pressure using a Model 500 Avco plasma generator. Temperatures were calculated from these sound speeds on the assumption that only the translational and rotational degrees of freedom were excited by the sound wave, while vibration, dissociation, and electronic excitation remained frozen. The temperatures determined in this way ranged from 3500 to 8000°K. In the case of the air arc, the ultrasonic temperature measurements were compared with temperatures determined from an energy balance on the plasma jet. The average temperatures obtained by the two methods agreed within 10% over the temperature and enthalpy range considered. The results also showed that temperature fluctuations as high as 50% can occur during a given arc run. A discussion of the experimental apparatus, method of measurement, experimental results and analysis will be given.

Session T. Physiological and Psychological Acoustics

IRA J. HIRSH, *Chairman*

Contributed Papers

T1. Influence of Different Acoustical Stimuli on the Threshold of the Contralateral Ear. M. LOEB AND J. L. FLETCHER, *U. S. Army Medical Research Laboratory, Fort Knox, Kentucky.*—A previous attempt to assess the attenuation produced by the intratympanic muscle reflex involved determination of the increase in absolute threshold following contralateral pure tone stimulation. This estimate was considerably smaller than estimates of reflex attenuation obtained by other means. In the present study a 110-db SL sine wave, square wave, narrow band noise, or broad band noise was introduced in one ear and the resultant threshold shift in the contralateral ear was measured. The smallest changes were produced by the square wave and pure tone stimuli. Appreciably larger threshold increases were elicited by the broad band and narrow band noises, especially the latter. Possible explanations of the obtained differences are discussed.

T2. Attenuation by the Acoustic Reflex as Indicated by Contralateral Remote Masking. W. Dixon Ward, *Research Center, Subcommittee on Noise, Los Angeles 57, California.*—One method of measuring the protection afforded by the muscles of the middle ear is to arouse the reflex by means of a loud sound in one ear and measure the concomitant threshold shift in the other. Operationally this amounts to the determination of contralateral remote masking (CRM): "remote" indicates that the arousal stimulus is somewhat higher in frequency than the test tone. When a steady noise is the arousal stimulus, the following phenomena are observed: (1) the CRM adapts steadily, reaching a stable value about 2 min after onset of the noise; (2) a mere change in frequency or level of the arousal stimulus is not sufficient to restore the CRM to a nonadapted level; (3) a 30-sec silent period will restore full CRM after adaptation; (4) maximum protection occurs at 500 cps; (5) medium-frequency octave and crossed-filter bands are somewhat more efficient than higher bands in producing CRM; (6) CRM increases linearly from 85 to 125 db SPL, reaching as high as 40 db in some individuals. Pure tones do not produce a significant sustained CRM. The results are shown to agree with recent animal studies by Wersäll and by Galambos and his associates. (This research was supported by grants B-1122 and B-2199 from NINDB, Public Health Service.)

T3. Cochleo-Tympanic Reflex Tests with Controlled Acoustic Stimuli. E. S. Mendelson, *Air Crew Equipment Laboratory, Naval Air Material Center, Philadelphia 12, Pennsylvania.*—In repeated tests on one subject, drumhead retractions were recorded from one ear while the other ear was stimulated acoustically. The amplitude and the spectral constitution of the stimuli were varied; the rise time for each stimulus remained constant. With the following stimuli: masking noise, 128, 256, 512, 1024, 1448, 2048, 2896, 4096, 5792, and 8192 cps, the apparent reaction thresholds averaged about 105, 115, 120, 110, 110, 110, 100, 95, 110, 105, and 115 db (re 0.0002 microbar), respectively. On different occasions average threshold estimates varied within ±10 db, whereas the contours of the stimulus-response magnitudes at the individual frequencies tended to be parallel from day to day. In every experiment the relative peak magnitudes of the reactions to 128, 4096, 5792, and 8192 cps were smaller than those to midfrequencies and noise. The appearance of the recordings obtained with the individual stimuli also suggests a systematic variation with spectral quality. The present indications are, however, obscured by uncertain acoustic and biological complexities, and call for further study.

T4. Steady vs Intermittent Noise Bursts for Evoking and Maintaining Middle Ear Muscle Reflex Reactions. E. S. Mendelson and D. L. Kemmerer (nonmember), *Air Crew Equipment Laboratory, Naval Air Material Center, Philadelphia 12, Pennsylvania.*—Since middle ear muscle reflex contractions reduce sound transmission from the outer to the inner human ear, their safe and effective induction is a protective problem. There are reports that (1) the reflex threshold is lowest at audible midfrequencies; (2) persisting noise causes longer maintained contractions than does persisting pure tone; (3) noise reduces auditory acuity less than pure tone at the same SPL; and (4) diplacusis is more readily caused by pure tone than by broadband noise. Midfrequency noise therefore seems likely to be the safest and most effective stimulus for intentional, protective activation of middle ear muscle reflexes. Cochleo-tympanic tests have been conducted upon two subjects, with noise, essentially flat from 1200 to 4800 cps, alternating short bursts of intermittent and steady stimulation at the same nominal SPL's, to estimate the practicability of reducing the total duration of the stimulus without positive loss of its effectiveness. Limiting repetition rates at which intermittent stimuli appeared as effective as steady stimuli for eliciting and maintaining the reflex reactions were found to vary with the SPL of the stimulus.

T5. Temporary Hearing Losses for Bare and Protected Ears as a Function of Exposure Time to Continuous and Firing Noise. Alexander Cohen, *Quartermaster Research and Engineering Command, U. S. Army, Natick, Massachusetts.*—Temporary hearing losses for frequencies 250–8000 cps were noted for bare and protected ears (tanker's helmet) following 6-, 12-, and 18-min exposures to recorded 30-cal machine-gun fire and continuous wideband noise of comparable total energy. Threshold losses for both types of noise were generally confined to tones above 1000 cps and tended to become greater with increasing exposure time. Continuous noise caused greater losses than the firing noise under bare ear conditions for the three exposure times. A comparison of these losses against those noted when using the tanker's helmet indicated that the helmet gave significant protection against continuous noise but little protection against the firing noise. A second experiment studied the recovery of 4000 cps threshold losses for a 20-min period following exposure to the noise conditions cited above. Especially for the longer exposure times, rates of threshold recovery from bare ear exposures to continuous noise were slower than those for the firing noise. As compared with the data for bare ears, the tanker's helmet gave faster rates of recovery from losses due to continuous noise but did not facilitate recovery of losses due to firing noise.

T6. Damage Risk Criterion for and Auditory Fatigue from the Audiac. K. D. Kryter, A. Z. Weisz (nonmember), and F. M. Wiener, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Approximately three years ago Dr. Wallace J. Gardner and Dr. J. C. R. Licklider invented a device, later named the *Audiac*, which apparently has certain analgesic effects when used in dentistry. For this purpose the *Audiac* provides to the patient's ears wide band noise and music. The intensity levels of the noise and music are, within certain limits, under the control of the patient. Before the first production model of the Audiac was made generally available, instructions and advice for its use with regard to possible damage risk to hearing were written. These instructions, based on a specified spectrum for the noise to be provided by the Audiac, were in part aimed at the prevention of an exposure to the noise and music beyond a specific damage risk criterion. This criterion was based on scientific data appearing in the medical, physiological and acoustical literature on the subject of temporary and permanent hearing loss due to exposure to sound. The purposes of the present paper are (1) to review briefly this damage risk criterion, (2) to present the results of laboratory tests conducted to measure the auditory fatigue caused by the noise and music from the Audiac, and (3) to compare the amount of temporary auditory fatigue observed in our tests and the fatigue one would predict according to our criterion, and according to recent mathematical formulas derived for this purpose by Ward, Glorig, and Sklar [J. Acoust. Soc. Am. 31, 522–528 (1959)].

T7. Psychopathologic Mechanisms in Aversive Responses to Noise. PETER F. OSTWALD, *University of California School of Medicine and The Langley Porter Neuropsychiatric Institute, San Francisco 22, California.*—The purpose of this presentation is to specify certain forms of breakdown in the human "survival-scheme" which, in combination with acoustic factors, can make noise intolerable. Twelve mentally-ill persons, ranging in age from 14 months to 80 years, and 7 normal volunteers were studied, using Wilmer's Auditory Projection Technique [Science (1951)] and other tape-recorded stimuli. Listener responses were studied in terms of verbalized labels for the imagined acoustic event, ideas about the internally felt impact of the sound, and statements about externally observable behavior. Dreams and hallucinations were noted whenever possible. The results were analyzed within a theoretical framework proposed by D. M. Green [J. Acoust. Soc. Am. (1960)], viz., that the listener's "criterion-level" is a significant determinant in decision-making phases of the threshold process. Assuming that listeners scan their environment for meaningful auditory signals, a noise seems to become disturbing when it simultaneously stimulates and frustrates the listener's search for nourishment, protection, contact, and pleasure. The intensity of each of these needs and the completeness with which it can be satisfied depend on multiple genetic, personal, and social factors. Therefore, it is suggested that effective medical noise-desensitization methods might be developed to augment the already successful acoustic approaches to noise control.

T8. Effect of Auditory Fatigue on Differential Intensity Thresholds. DONALD N. ELLIOTT, WINIFRED RIACH (nonmember), AND HERMAN SILBIGER, *Wayne State University, Detroit 2, Michigan.*—Although the great majority of studies on auditory fatigue have utilized the TTS (temporary threshold shift) as the primary index of fatigue, it also manifests itself in many other ways. Several studies have reported the existence of recruiting in the fatigued ear—as demonstrated at suprathreshold intensities by the use of loudness balance tests, and at threshold intensities by a decrease in the size of the recording attenuator pen excursions when the absolute threshold recovery curve is traced. This paper is a report of the change in the size of the intensity dl's following fatigue. Two parameters have been investigated, viz., the severity of the fatigue as defined by the TTS at the time of the dl determinations, and the sensation level (post stimulation) of the test tone. Both of these factors are found to be related to the change in the dl's.

T9. Shifts in the Masked Threshold. IRA J. HIRSH, *Central Institute for the Deaf, St. Louis 10, Missouri.*—Unpublished observations by Michel Burgeat in 1958 have now been repeated and extended. When the typical masked threshold of a tone is measured in the presence of a white noise, it is very stable over time. When, however, remote masking is measured for low-frequency tone in the presence of a high-frequency band of noise, initial threshold just after the noise is turned on is higher than it is subsequently. This reduction in remote masking takes place in from 1 to 3 min depending upon the noise level. The amount of shift varies from 1 or 2 to as much as 10 or 12 db. A similar shift is shown even for the threshold of the low-frequency tone measured on the ear opposite to the one receiving the noise. This second observation casts some doubt on the relevance of the remote-masking process for this shift. Interpretation involving the acoustic reflexes will be discussed.

T10. The Effect of Matching Time on Perstimulatory Auditory Adaptation. ARNOLD M. SMALL, JR., AND FRED D. MINIFIE (nonmember), *University of Iowa, Iowa City, Iowa.*—If perstimulatory adaptation is measured in terms of a loudness balance between the ear (experimental) undergoing sustained stimulation and the contralateral (comparison) ear, it has been assumed that the shorter the duration of the comparison stimuli the more accurately the loudness balances will reflect adaptation in the experimental ear. That is, the shorter the comparison stimuli the smaller the amount of adaptation in the comparison ear itself. This study attempted to provide information regarding the nominal perstimulatory adaptation when both the duration and the interval between comparison stimuli were varied systematically. The experimental stimulus consisted of a 4000 cps tone at 80 db SL. The comparison stimulus was a tone of identical frequency derived from the same oscillator whose intensity was controlled by the listeners through a motor driven attenuator. Both the *on* and *off* times of the comparison stimulus were varied from 10 to 60 sec in 10-sec steps. The results indicate that as either the comparison stimuli are shortened or interstimulus interval lengthened, the amount and rate of measured adaptation increase. The maximum adaptation rate approximates 20 db/min and occurs immediately following initiation of the experimental stimulus. Under most conditions adaptation reaches an asymptote by about 6 min, the maximum amount being about 30 db. Under equivalent conditions, changes in *on* time produce greater changes in rate and amount of adaptation than do similar changes in *off* time.

T11. Age and Sex Differences in Pure-Tone Thresholds. Addendum I: Age Group 51 to 57 Years. JOHN F. CORSO, *Department of Psychology, The Pennsylvania State University, University Park, Pennsylvania.*—The present paper is an addendum to a laboratory study published in 1959 on the normal thresholds of hearing for pure tones for an age-stratified sample of subjects drawn from a population exposed to minimal levels of industrial noise. It provides audiometric data (250–8000 cps) for 205 subjects in a 51- to 57-year-old group to supplement the earlier data for 500 subjects in four age groups between 18 to 49 years. The results of this study confirm the previously reported findings that (1) there is less than 5 db difference in the average sensitivity between right and left ears, (2) there is a decrease in hearing sensitivity with advancing age for both men and women, and (3) hearing loss with advancing age spreads progressively from higher to lower frequencies. Although women in general have better hearing and less intersubject variability than men, this trend is reversed for the lower frequencies in the 51- to 57-year-old age group. It is concluded that while the onset of presbycusis is later for women than for men, auditory deterioration proceeds at a faster rate in women than in men after about 50 years of age.

Session U. Shock and Vibration Acoustics

HORACE M. TRENT, *Chairman*

Contributed Papers

U1. Linearity in Nonlinear Response and Its Application to Acoustical Fatigue Investigations. PAUL WANG, *North American Aviation, Inc., Los Angeles 45, California.*—Experimental studies were performed on simple structural elements subjected to sinusoidal and random noise excitations in both linear and nonlinear response regions. On the basis of results obtained, a theory is formulated to account for the observed nonlinearity and to explain the possible factors causing sudden amplitude changes inherent in skewed response peaks identifiable to nonlinear cases. By using familiar resonance equations of one degree of freedom, an alternative approach is offered and compared with current understandings of acoustically induced stresses under high level excitation. Sample results will be employed to illustrate the feasibility of applying these findings to fatigue life expectancy calculations for random loading. The methods used in determining damping coefficient ratio will also be discussed.

U2. Transient Behavior of Nonlinear Isolation Mountings. J. C. SNOWDON, *Ordnance Research Laboratory, Pennsylvania State University, University Park, Pennsylvania.*—The response of damped nonlinear isolation mountings to step-like foundation displacements has been determined theoretically. Both "stiffening" and "softening" mounts have been considered. Utilization of the techniques of numerical integration (performed by a digital computer) rather than perturbation methods has made possible the consideration of large departures of mount stiffness from linearity. For brevity, the results presented relate only to a value of mount damping (linear) equal to five percent of critical damping. For the majority of step rise times considered, the softening mount is shown to reduce both the acceleration and the displacement of the mounted item below the values observed for a linear or a stiffening mount. The reduction in acceleration can be comparable with 10 db, and it is frequently the magnitude of the acceleration to which the likelihood of damage to the contents of the mounted item may be related. (This investigation was conducted at the University of Michigan, Ann Arbor, Michigan, under the sponsorship of the U. S. Navy, Bureau of Ships.)

U3. Random Fatigue of Hard Spring Resonators. P. W. SMITH, JR., *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—Computations have been made for the effect of a nonlinearity of the hard spring type upon the fatigue damage ensuing on excitation of a resonator by a stationary, broad-band, random Gaussian force. The fatigue theory used assumes a simple accumulation of damage in proportion to a power of the peak values of response strain, without interactions between peaks. Two cases are distinguished: (a) strain proportional to displacement; (b) strain nonlinearly dependent on displacement, as is the surface fiber strain in a bar with pinned ends. Numerical results are presented that indicate for case (a) a continuous decrease, with increasing nonlinearity, of the rate of accumulation of damage, while for case (b) this damage rate increases to a maximum, as large as 3.4 times the linear estimate, and then falls below the value for linear response. Distortion of the curves for peak-strain density and for fatigue-damage density is discussed. (Research sponsored by Air Research and Development Command, U. S. Air Force.)

U4. Narrow Band Excitation of the Hard Spring Oscillator. RICHARD H. LYON AND MANFRED HECKL, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—By using the technique of "quasi-linearization," the response of a hard-spring oscillator to a band of random noise is calculated. Comparison is sought with the limiting cases of wide band (purely random) and determinate sinusoidal excitation. It is shown that the limiting cases are approached in a satisfactory manner. In particular, multivalued solutions are obtained as the bandwidth of excitation is diminished but do not occur when the bandwidth is suitably broad. Experimental illustration is included. (Supported in part by the Office of Naval Research and the National Science Foundation.)

U5. Evaluation of Dynamic Characteristics of Resilient Materials. I. P. VATZ, *General Dynamics Corporation, Electric Boat Division, Groton, Connecticut.*—The problem of experimentally determining the dynamic characteristics of resilient vibration isolation materials has been solved using two techniques. The test methods, instrumentation and the readouts in terms of impedance and transmissibility are described. Results of impedance and resonance analysis are discussed for several sample materials under different degrees of loading. The results are compared with theoretical values. Static values for materials with high hysteresis are also given in terms of standard stress-strain curves. The differences between static and dynamic spring constants are discussed.

U6. Combined Bending—Torsion of a Closed Beam with the Addition of Axial Loading. HERBERT SAUNDERS (nonmember), *General Electric Company, Missile and Space Vehicle Department, Philadelphia 4, Pennsylvania.*—The basic Myklestad method for the determination of combined bending-torsion loads on a beam is supplemented by the addition of axial loading. The beam is considered to consist of finite lumped masses and their associated springs. The procedure employed is the "transfer matrix" approach making the technique readily applicable to the digital computer. The author extends a note originally published by him [J. Acoust. Soc. Am. 32, 409 (1960)] concerning the addition of rotary inertia and shear deformation. As examples, the author determines (1) the buckling load of a static column and (2) the critical length to prevent buckling of a thin column loaded by its own weight.

U7. Power Flow between Structural Modes and Reverberant Fields. RICHARD H. LYON, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The response of a resonant structure in a reverberant noise field is viewed as the attempt of a set of damped oscillators to come to equilibrium with a thermal bath. The asymptotic (in time) energy in the modes is evaluated from a simple energy equivalent circuit and depends on the ratio of internal to radiation resistance and the "temperature" of the sound field. The connection between these results and those of P. W. Smith, Jr. [J. Acoust. Soc. Am. 32, 929(A) (1960)] is demonstrated. Experimental illustration of the method is included. (Supported in part by Wright Air Development Division.)

U8. Some Investigations on Grillage Vibrations. MANFRED HECKL, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—The vibrations of grillages consisting of beams crossing each other at right angles were investigated theoretically and experimentally. It was found that at low frequencies grillages behave like orthotropic plates. At high frequencies (when the distance between the intersections is much larger than the beam bending wavelength) more complicated vibration patterns are observed. For the special case of point excitation, a certain highly frequency dependent attenuation occurs at the intersections closest to the source. No additional attenuation could be measured at some distance from the source, due to excitation of torsional waves at the intersections. (Supported by the Office of Naval Research.)

U9. Damping of Flexural Waves by "Patches" of Treatment. R. J. McQuillin and E. M. Kerwin, Jr., *Bolt Beranck and Newman Inc., Cambridge 38, Massachusetts.*—The damping of vibrating systems by uniform applications of damping treatments has been discussed earlier [J. Acoust. Soc. Am. **31**, 952 (1959); **32**, 912(A) (1960); **32**, 1513(A) (1960)]. However, in some practical applications only a part of the system may be available for treatment. The question of partial coverage is also raised when treatment weight is an important consideration. The present studies present the first step toward describing the effectiveness of partial treatments or "patches." We consider a system (beam or plate) having a one-dimensional standing flexural wave of the form A coskx. We consider both free-viscoelastic-layer and constrained-viscoelastic-layer damping treatments. Damping effectiveness is evaluated as a function of patch length, patch position in the standing-wave field, shear parameter, viscoelastic material loss factor, and constraining-layer boundary conditions. The concept of a damping efficiency for the patch is introduced. Thus, for a given system the damping can be evaluated as the product of the fractional surface coverage and the damping efficiency. (Supported by the U. S. Navy, Bureau of Ships.)

U10. Some Structural Design Features of the WADD Sonic Fatigue Test Facility. Donald J. Neubauer (nonmember), *Daniel, Mann, Johnson and Mendenhall, Washington 6, D. C.*, Lloyd J. Williams (nonmember), and Eric E. Ungar (nonmember), *Bolt Beranck and Newman Inc., Cambridge 38, Massachusetts.*—Construction of a large high-intensity sonic fatigue facility is under way at Wright-Patterson Air Force Base, Dayton, Ohio. No previous large structure has been required to withstand long term intense excitation at all frequencies in a wide band, so that structural design of this facility was faced with some unprecedented problems. The present paper outlines the test chamber size and performance requirements and how these led to the present design. The present structure is described, and design of viscoelastically damped configurations is discussed. The sizes and design stress levels of typical beams and panels in this structure are compared to those encountered in conventional structures. Some interesting details of the structure are pointed out.

U11. Vibrations in a Large Softly Clamped Glass Plate. James N. Lange, *The Pennsylvania State University, University Park, Pennsylvania.*—The 3×2.5 square-meter glass plate was driven electrodynamically at the center and the amplitude distribution as a function of the frequency was recorded for positions out to and along the edge of the plate. The loss factor was measured by both bandwidth and decay methods. Transients were excited by a shot from a blank pistol and photographs of the resulting oscilloscope pattern were taken. (This work was sponsored by the U. S. Atomic Energy Commission and Bureau of Naval Weapons.)

U12. Noise and Vibration Measurements of a Large Ventilating Fan for Various Degrees of Vibration Isolation. Laymon N. Miller, *Bolt Beranek and Newman Inc., Cambridge 38, Massachusetts.*—We were recently presented with the opportunity of making a series of noise and vibration measurements on a 25-hp ventilating fan located above an occupied floor in an office building. The measurements were carried out during the process of trying to diagnose the cause of excessive noise levels in the office floor beneath the fan. Vibration levels at various parts of the fan base and noise levels in the office area below are presented for various degrees of isolation of the fan on its mounting.

Session V. Musical Acoustics

Daniel W. Martin, *Chairman*

Contributed Papers (20 min.)

V1. The Hamograph: A New Amplitude-Rhythm Control Device for the Production of Electronic Music. Myron Schaeffer (nonmember), *Electronic Music Studio, University of Toronto, Toronto, Ontario, Canada.*—The production of electronic music forces the composer to tackle certain new problems which require unusual and often difficult technical solutions. Some of these are the synchronization of melodic units into a contrapuntal texture; the organization of sounds into complex rhythmic continuities; shaping individual sounds in relation to attack, continuum, and decay patterns; exact repetition and/or with tempo variations of rhythmic patterns with or without alteration of pitch. The principle of the Hamograph is that of drawing with conductive ink or of pasting metal foil contours on 35-mm continuous loops which are read by a series of silver brushes so arranged that the signal circuit is modified in intensity by means of a resistor chain which in turn controls the volume of the final signal in the aural monitor or recording device. This device provides a simple means of organizing multiple sound sources into useful musical objects by (1) creating and repeating complex rhythmic patterns without splicing; (2) creating complex rhythmic montages without splicing or transfer; (3) repeating exact envelope patterns with same or varying timbre; (4) programming sections of a composition using prepared sound objects without transfer; (5) changing the tempo of rhythmic organizations without change of pitch; and (6) modifying elements of the control plan without reconstructing the total control system.

V2. New Tool for the Exploration of Unknown Sound Resources for Composers. Harald Bode (nonmember), *The Wurlitzer Company, Corinth, Mississippi.*—When applying suited means in order to modify known natural or electronically created sounds, more or less unusual and some very usable new phenomena may be originated. The methodic exploration, evaluation, and exploitation of these will be rewarding for the composer, who is searching for new resources. As a tool for these purposes, a new modular sound and envelope shaping, modulating, and modifying device has been developed, which may be combined as an integral unit with an alternating sound pattern creating apparatus. These devices may be applied in many ways to modify one or several audio phenomena as well as by using in a predetermined way one feature to affect another. These systems comprise an arbitrary selection of modules, some of which are well known in audio and communication techniques, but many combinations of which represent unusual systems with interesting functions. A relatively simple version of this type of modular assembly with a surprising variety of performance features has been built and used for experimental purposes. It will be briefly described here as a whole and in some of its details; a selection of significant tape recordings will be demonstrated.

V3. Aid to Music Composition Employing a Random-Probability System. Harry F. Olson and Herbert Belar (nonmember), *RCA Laboratories, Princeton, New Jersey.*—The art of music composition can be learned to some extent by following the work of composers and by studying the fundamentals of music. Composition can be stimulated by various experiences and may be aided by research. However, the process which operates in the mind of the composer is not understood. Even without such knowledge, it is possible to develop machines which can aid the composer in his search for melody which is the essence of most music. Although the ability to create is a gift, even great composers have at times turned to various aids for assistance. In this connection, random-probability systems allow the generation of sequences of notes in a series which are not completely random nor completely ordered, but each note is selected in a random fashion with a probability which depends upon preceding notes. Such a machine has been developed and has been operated to generate a sequence of an almost infinite series of notes according to trinote probabilities found in the music of Stephen Foster. Combined with sound synthesizers, the machine provides additional instrumentation for the musical laboratory. However, the ultimate output of even the best equipped laboratory still depends upon the composer.

V4. Excitation of String Instruments. Melville Clark, Jr., Daniel Schwartz (nonmember), and William Feldman (nonmember), *Massachusetts Institute of Technology, Cambridge 39, Massachusetts.*—Stroboscopic lights and cameras are utilized to photograph the excitation and motion of piano, violin, and harp strings; high speed pictures of which will be shown. Fingers after plucking harp strings are not hit by the string, the shape of which may be easily computed. The wave character may be easily seen from the persistence of waveshape. Piano hammers hitting wound strings oscillate in the plane of the shank and hammer. The contact time (2–7 msec) is relatively independent of the speed of depression and frequency of the note (C_1–C_5). The piano action itself works much faster than it could be played. Very slight painting of a wound string greatly damps the sound. Terminal hammer speeds vary from 30 to 600 msec. The bow pulls the string aside without slipping. The return of the previously created pulse, reflected and inverted at the nut, snaps the string away from the bow in $\frac{1}{8}$ of a period. The string deflects an equal and opposite amount before being picked up and replucked by the bow again. (This work was performed in Professor Harold Edgerton's laboratory and was supported by the National Association of Music Merchants.)

V5. Influence of the Reed on the Vibration Frequency of Clarinet Tones. John Backus, *University of Southern California, Los Angeles 7, California.*—The frequency of a clarinet tone is influenced to some extent by the lip pressure on the reed, giving the clarinettist some control over the intonation of the instrument. The resonant frequencies of the instrument measured by external excitation with the reed aperture closed are some 30–50 cents higher than the actual playing frequencies, which in turn can be varied over a range of about 50 cents. A small vibration theory of this effect has been developed based on the observation that for low intensity tones the pressure variation in the air column is practically sinusoidal [J. Acoust. Soc. Am. 32, 1493 (1960)]. A velocity potential appropriate to a vibrating air column with wall damping is assumed and is evaluated to give the required volume flow through a reed aperture whose width is varying sinusoidally with time. The result shows a lowering of frequency from that of a completely closed tube by the fractional amount $\omega M/2RQ$, where Q is that for the clarinet, and M and R are the mass and resistance for volume flow through the average reed aperture. Values of M and R have been calculated theoretically and checked experimentally, with good agreement; these values used in the formula above give results in fair agreement with observed frequency shifts. The existence of a threshold blowing pressure is also explained. (This work was supported by the National Science Foundation.)

V6. On Defining the Range of Pitch Perception. Newman Guttman, *Bell Telephone Laboratories, Inc., Murray Hill, New Jersey.*—In stating that pitch is an attribute which may be ordered on a high-low scale, the American Standards definition refrains from setting frequency limits to pitch perception. This may be due to fear that presence of harmonics arising out of distortion of low-frequency sinusoids makes responses in this region incompatible with responses to high-frequency sinusoids. We suggest that inasmuch as 200–400-cps unipolar pulse trains with or without fundamental component produce essentially the same pitch perceptions as sinusoids, such trains may be used to examine low pitches without frequency restrictions. Perhaps then it becomes justifiable to confine pitch perception by definition to a range from the highest perceivable frequencies to the low frequencies where the high-low dimension slides into the fast-slow dimension. It may furthermore be useful to revive the idea of defining an extent of musical pitch within the over-all range. We might assign the presence of the octave as the criterion of this extent. The upper limit of musical pitch is about 4500 cps, and our tests of the octave with pulse trains place the lower limit at about 35 cps.

V7. Measurement of the Musical Interval Sense. Andrew G. Pikler (nonmember), and J. Donald Harris, *U. S. Naval Medical Research Laboratory, Groton, Connecticut.*—Ten generally unmusical *Ss* were used in an effort to dissect and quantify the hierarchy of cues which enable the trained instrumentalist or singer to create a progression of musical intervals with great precision, speed, and consistency. Experiment A required *Ss* to create one at a time, by adjustments, the eight intervals of the major scale (C fixed at 262 cps). Errors were commonly half a semitone. The Octave was almost always significantly and seriously sharp. Evidently the intervals in isolation sustain an average error of up to a quartertone. Experiment B required *Ss* to set seven oscillators arranged in a rudimentary electric piano to create the scale. The intervals showed improvement in placement and variance, though deviations were still commonly a fourth of a semitone. Evidently the context of even a very primitive musical line is a distinct additional cue in pitch intonation. Experiment C was as B except that the tonic and octave were both fixed. With this additional anchor, placement of all tones was improved; for example, the fifth improved from 48 cents sharp to 21.5 cents sharp. Evidently errors increase with distance from an anchor. The further cue of harmony, not studied here, would suffice to bring intonation errors within limits of the DL.